EAST Search History

S6	50	S5 and updat$4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/02/04 13:49
S7	8	S2 and (minimum mean square error)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/02/04 13:58
S8	513	fixed codebook	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/07/22 20:04
S9	79	S8 and (mean square error)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/07/22 20:04
S10	76	S9 and vector	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/07/22 20:04
S11	51	S10 and updat$4	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/07/22 20:04
S12	44	S11 and initial$6	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/07/22 20:04
S13	44	S9 and (initial$6 with vector)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/07/23 11:38
S14	39	S13 and updat$6	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/07/22 20:51
S15	532	itu with "g.729"	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/07/22 20:53

EAST Search History

S16	8	S10 and S15	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/07/22 20:53
S17	14	(initializ$4 adj2 vector) and (fixed codebook)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/07/23 11:45
S18	15	(modif$5 adj2 vector) and (fixed codebook)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/07/23 11:45
S19	4	(modif$5 adj2 vector) with (fixed codebook)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2006/07/23 11:46
S20	0	(initializ$5 near vector) and (fixed codbook)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2007/03/16 12:00
S21	1824	(initializ$5 near vector)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2007/03/16 12:01
S22	16	S21 and acelp	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2007/03/16 12:01
S23	588	fixed codebook	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2007/03/16 12:13
S24	12	S23 and (initializ$4 vector)	US-PGPUB; USPAT; EPO; JPO; DERWENT; IBM_TDB	ADJ	ON	2007/03/16 12:13

[File 2] **INSPEC** 1898-2007/Apr W1

[File 6] **NTIS** 1964-2007/Apr W1

[File 8] **Ei Compendex(R)** 1884-2007/Mar W4

[File 34] **SciSearch(R) Cited Ref Sci** 1990-2007/Apr W1

[File 35] **Dissertation Abs Online** 1861-2007/Mar

[File 56] **Computer and Information Systems Abstracts** 1966-2007/Mar

[File 57] **Electronics & Communications Abstracts** 1966-2007/Mar

[File 65] **Inside Conferences** 1993-2007/Apr 06

[File 95] **TEME-Technology & Management** 1989-2007/Apr W1

[File 99] **Wilson Appl. Sci & Tech Abs** 1983-2007/Mar

[File 144] **Pascal** 1973-2007/Apr W1

[File 239] **Mathsci** 1940-2007/May

[File 256] **TecInfoSource** 82-2007/Oct

[File 434] **SciSearch(R) Cited Ref Sci** 1974-1989/Dec

[File 583] **Gale Group Globalbase(TM)** 1986-2002/Dec 13

**File 583: This file is no longer updating as of 12-13-2002.*

[File 603] **Newspaper Abstracts** 1984-1988

**File 603: This is a closed file.*

[File 483] **Newspaper Abs Daily** 1986-2007/Apr 08

[File 248] **PIRA** 1975-2007/Mar W2

Set Items Description

```
S1      279600    S (VOICE OR SPEECH OR SOUND OR NATURAL()LANGUAGE?? OR WORD OR
TEXT OR LEXIC? OR PHRASE? OR TEXT()STRING??)(3N)(SYNTHESI? OR GENERAT??? OR
RECOGNI? OR PROCESS? OR SEGMENT?? OR PORTION?? OR PART?? OR COMPONENT?? OR
SPETRA OR SPECTRUM OR BAND?? OR CONCAT?)
S2        2883    S PITCH(3N)(PREDICT??? OR CONTOUR?? OR INTONATE??? OR
MODULAT???)
S3         164    S (CALCULAT? OR ESTIMAT??? OR DETERMIN? OR PROBABILIT??? OR
RANDOM OR STATISTIC??? OR STOCHASTIC)(3N)S2
```

```
S4        2538    S (LOW()FREQUENC???)(3N)(ENERGY OR ENERGIES OR ENTROP?)
S5          86    S (INCREAS??? OR ADD?? OR IMPROV??? OR OPTIMI?)(3N)S4
S6           4    S FILTER??(3N)(S4 OR S5)
S7         349    S AU=(EIDE, E? OR EIDE E? OR BAKIS, R? OR BAKIS R?)
S8           0    S (S2 OR S3)(3N)(S4:S6)
S9           2    S (S2 OR S3)(S)(S4:S6)
S10          2    S (S2 OR S3) AND (S4:S6)
S11          0    S S10 NOT S9
S12         79    S (S2 OR S3)(3N)FILTER???
S13          0    S S12(3N)(LOW()FREQUENC???)(3N)(ENERGY OR ENERGIES OR ENTROP?)
S14          0    S S12(S)(LOW()FREQUENC???)(3N)(ENERGY OR ENERGIES OR ENTROP?)
S15          0    S S12 AND (LOW()FREQUENC???)(3N)(ENERGY OR ENERGIES OR ENTROP?)
S16          0    S S12(20N)(LOW OR FREQUENC???)(3N)(ENERGY OR ENERGIES OR
ENTROP?)
S17         10    S S12(20N)(CALCULAT? OR ESTIMAT??? OR DETERMIN? OR
PROBABILIT??? OR RANDOM OR STATISTIC??? OR STOCHASTIC)
S18          6    RD  (unique items)
S19          4    S (S2 OR S3 OR S12) AND S7
```

9/3,K/1 (Item 1 from file: 144) **Links**
Pascal

13555872 PASCAL No.: 98-0257511

Amplitude modulation cues for perceptual voicing distinctions in noise

STROPE Brian P; ALWAN Abeer
Dept. of Elec. Eng., UCLA, 66-147E Eng. IV, 405 Hilgard Ave., Los Angeles, CA 90095
Journal: The Journal of the Acoustical Society of America
, 1998-05, 103 (5
) 2771-2772
Language: English

... vocal folds can modulate the pressure source behind the constriction sufficiently to generate an acoustic **pitch**-rate amplitude-**modulation** cue in high-frequency regions. Predictions of the perceptual experiment<right single quotation mark>s...

9/3,K/2 (Item 2 from file: 144) **Links**
Pascal

11756456 PASCAL No.: 94-0627433

Effect of fundamental frequency perturbations on medial stop-consonant (voice) judgments

MOLIS Michelle R; DIEHL Randy L
Dept. of Psychol., Univ. of Texas at Austin, Austin, TX 78712
The 128th meeting of the Acoustical Society of America (Austin, Texas

(USA)) 1994-11-28/1994-12-02
Journal: Journal of the Acoustical Society of America
, 1994-11, 96 (5
) 3228-3228
Language: English

... synthesized by varying VOT from 10 to 45 ms in 5 ms steps. Fifteen different **pitch contours** were generated by designating F0 targets at three points in the stimulus: initial vowel, onset...

18/3,K/1 (Item 1 from file: 2) **Links**
INSPEC

09349588 **INSPEC Abstract Number:** B2005-05-6130-036
Title: Adaptive two-sided pitch prediction filters for pathological voices analysis
Author Kacha, A.; Grenez, F.; Bettens, F.; Schoentgen, J.
Author Affiliation: Dept. Waves & Signals, Univ. Libre de Bruxelles, Brussels, Belgium
Conference Title: Proceedings of International Conference on Signals and Electronic Systems ICSES'04 p. 417-20
Editor(s): Bartkowiak,M.; Domanski,M.; Grajek,T.; Stasinski,R.; Swierczynski,R.; Rosinski,T.
Publisher: Polish Society for Theoretical and Applied Electrical Engineering , Poznan, Poland
Publication Date: 2004 **Country of Publication:** Poland xvii+603 pp.
ISBN: 83 906074 7 6 **Material Identity Number:** XX-2005-00450
Conference Title: Proceedings of International Conference on Signals and Electronic Systems ICSES'04
Conference Date: 13-15 Sept. 2004 **Conference Location:** Poznan, Poland
Language: English
Subfile: B

Abstract: ...correlates with the perceived degree of hoarseness. The coefficients of the time-varying two-sided **pitch prediction filter** are **estimated** adaptively by means of a recursive least squares algorithm.

18/3,K/2 (Item 2 from file: 2) **Links**
INSPEC

07748455 **INSPEC Abstract Number:** B2000-12-6130C-046
Title: Pitch-synchronous linear-prediction analysis by synthesis with reduced pulse densities
Author Guerchi, D.; Qian, Y.; Mermelstein, P.
Author Affiliation: INTS-Telecommun., Univ. du Quebec, Verdun, Que., Canada
Conference Title: 2000 IEEE International Conference on Acoustics, Speech, and Signal Processing. Proceedings (Cat. No.00CH37100) **Part** vol.3 p. 1491-4 vol.3
Publisher: IEEE , Piscataway, NJ, USA
Publication Date: 2000 **Country of Publication:** USA 6 vol. lxxx+3906 pp.
ISBN: 0 7803 6293 4 **Material Identity Number:** XX-2000-01776
U.S. Copyright Clearance Center Code: 0 7803 6293 4/2000/$10.00
Conference Title: Proceedings of 2000 International Conference on Acoustics, Speech and Signal Processing
Conference Sponsor: IEEE; Signal Process. Soc
Conference Date: 5-9 June 2000 **Conference Location:** Istanbul, Turkey
Language: English
Subfile: B

Abstract: ...for techniques that reduce the perceptibility of the errors in the reconstructed signal. Pitch-synchronous **estimation** of the linear- **prediction filter** and **pitch**-synchronous updating of the adaptive codebook reduce the coefficient-**estimation** error and increase the relative contribution of the adaptive codebook component to the synthesized signal...

18/3,K/3 (Item 3 from file: 2) **Links**
INSPEC

05329962 **INSPEC Abstract Number:** B9303-6130-005, C9303-5260S-002
Title: Linear and nonlinear adaptive filtering and their application to speech intelligibility enhancement
Author Gu, Y.H.
University: Eindhoven Univ. Technol., Netherlands
Publisher: Eindhoven Univ. Technol , Eindhoven, Netherlands
Publication Date: 15 Sept. 1992 **Country of Publication:** Netherlands 218 pp.
Language: English
Subfile: B C
Abstract: ...noise itself is interference-speech as well. For this purpose, new linear and nonlinear adaptive **filtering** techniques and robust **pitch contour estimation** algorithms were developed and applied to co-channel speech separation. These new adaptive **filtering** algorithms and the **pitch contour estimation** algorithm are suitable for other applications apart from speech intelligibility enhancement.

18/3,K/4 (Item 1 from file: 8) Links
Ei Compendex(R)

11085841 **E.I. No:** EIP06269958251
Title: Harmonic filtering for joint estimation of pitch and voiced source with single-microphone input

Author: Lee, S.W.; Soong, Frank K.; Ching, P.C.
Corporate Source: Department of Electronic Engineering Chinese University of Hong Kong, Hong Kong, Hong Kong
Conference Title: 9th European Conference on Speech Communication and Technology
Conference Location: Lisbon, Portugal **Conference Date:** 20050904-20050908
E.I. Conference No.: 67499
Source: 9th European Conference on Speech Communication and Technology 9th European Conference on Speech Communication and Technology, Eurospeech Interspeech 2005.
Publication Year: 2005
Language: English
Abstract: ...the harmonic structure of voiced speech, pitch information of one source is extracted from the **pitch prediction filter** and the output residual becomes the **estimate** of the other source. The procedure is iterated successively with a summation constraint. From the evolution of **pitch prediction filter**, it is shown that the iterative harmonic filtering with the summation constraint is effective to...

18/3,K/5 (Item 2 from file: 8) Links
Fulltext available through: ScienceDirect
Ei Compendex(R)

07102398 **E.I. No:** EIP95032605386
Title: Fine pitch contour extraction by voice fundamental wave filtering method

Author: Ohmura, Hiroshi
Corporate Source: Electrotechnical Lab, Ibaraki, Jpn
Conference Title: Proceedings of the 1994 IEEE International Conference on Acoustics, Speech and Signal Processing. Part 2 (of 6)
Conference Location: Adelaide, Aust **Conference Date:** 19940419-19940422
E.I. Conference No.: 42612
Source: Proceedings - ICASSP, IEEE International Conference on Acoustics, Speech and Signal Processing v 2 1994. IEEE, Piscataway, NJ, USA,94CH3387-8. p 189-192
Publication Year: 1994
CODEN: IPRODJ **ISSN:** 0736-7791
Language: English
Identifiers: Fine pitch contour extraction; Voice fundamental wave **filtering** method; **Pitch** channel signal; **Pitch contour determination**; Wave segmentation

18/3,K/6 (Item 1 from file: 95) Links
Fulltext available through: USPTO Full Text Retrieval Options
TEME-Technology & Management

00937754 I95115783352

Harmonics estimation based on instantaneous frequency and its application to pitch determination of speech

(Oberschwingungsschaetzung auf Grundlage der augenblicklichen Frequenz und Anwendung auf die Tonhoehenbestimmung von Sprache)
Abe, T; Kobayashi, T; Imai, S
Precision & Intelligence Lab., Tokyo Inst. of Technol., Yokohama, Japan
IEICE Transactions on Information and Systems, vE78-D, n9, pp1188-1194 , 1995
Document type: journal article **Language:** English
Record type: Abstract
ISSN: 0916-8532
Identifiers: HARMONICS **ESTIMATION**; INSTANTANEOUS FREQUENCY; PITCH **DETERMINATION**; CENTER FREQUENCIES; PITCH EXTRACTION; PITCH **DETERMINATION** ALGORITHM; EXTRACTED **PITCH CONTOUR**; NONLINEAR **FILTERING**; PDA; Sprachsignal; Oberschwingung; Fourier-Analyse; Bandpass

19/3,K/1 (Item 1 from file: 2) **Links**
INSPEC

10142442
Title: Towards pooled-speaker concatenative text-to-speech
Author Eide, E.M.; Picheny, M.A.
Author Affiliation: IBM Thomas J. Watson Res. Center, USA
Conference Title: 2006 IEEE International Conference on Acoustics, Speech, and Signal Processing (IEEE Cat. No. 06CH37812C) p. I-73-6
Publisher: IEEE , Piscataway, NJ, USA
Publication Date: 2006 **Country of Publication:** USA CD-ROM pp.
ISBN: 1 4244 0469 X **Material Identity Number:** XX-2006-00798
U.S. Copyright Clearance Center Code: 1-4244-0469-X/06/$20.00
Conference Title: 2006 IEEE International Conference on Acoustics, Speech, and Signal Processing
Conference Date: 14-19 May 2006 **Conference Location:** Toulouse, France
Language: English
Subfile: C
Copyright 2006, The Institution of Engineering and Technology
Author Eide, E.M.; Picheny, M.A.
Abstract: ...to-speech system. First, we investigate the pooling of data from multiple speakers for building **statistical** models to **predict pitch** and duration, and present listening test results which show that the expressiveness of our TTS...

19/3,K/2 (Item 2 from file: 2) **Links**
INSPEC

08769140 **INSPEC Abstract Number:** B2003-12-6130E-125, C2003-12-5260S-071
Title: Recent improvements to the IBM trainable speech synthesis system
Author Eide, E.; Aaron, A.; **Bakis, R.**; Cohen, R.; Donovan, R.; Hamza, W.; Mathes, T.; Picheny, M.; Polkosky, M.; Smith, M.; Viswanathan, M.
Author Affiliation: IBM T. J. Watson Res. Center, Yorktown Heights, NY, USA
Conference Title: 2003 IEEE International Conference on Acoustics, Speech, and Signal Processing (Cat. No.03CH37404) **Part** vol.1 p. I-708-11 vol.1
Publisher: IEEE , Piscataway, NJ, USA
Publication Date: 2003 **Country of Publication:** USA 6 vol.(xcviii+927+852+788+883+823+764) pp.
ISBN: 0 7803 7663 3 **Material Identity Number:** XX-2002-01306
U.S. Copyright Clearance Center Code: 0-7803-7663-3/03/$17.00
Conference Title: Proceedings of International Conference on Acoustics, Speech and Signal Processing (ICASSP'03)
Conference Sponsor: IEEE Signal Process, Soc
Conference Date: 6-10 April 2003 **Conference Location:** Hong Kong, China
Language: English
Subfile: B C
Copyright 2003, IEE
Author Eide, E.; Aaron, A.; **Bakis, R.**; Cohen, R.; Donovan, R.; Hamza, W.; Mathes, T.; Picheny, M.; Polkosky, M.; Smith, M...
Abstract: ...led to significant gains in the output quality. On the algorithms side, we have introduced **statistical** models for **predicting pitch** and duration targets which replace the rule-based target generation previously employed. Additionally, we have...

Identifiers: ...**pitch prediction**;

19/3,K/3 (Item 1 from file: 8) **Links**
Fulltext available through: ScienceDirect
Ei Compendex(R)

09537136 **E.I. No:** EIP03397648837
Title: Recent improvements to the IBM trainable speech synthesis system

Author: Eide, E.; Aaron, A.; **Bakis, R.**; Cohen, P.; Donovan, R.; Hamza, W.; Mathes, T.; Picheny, M.; Polkosky, M.; Smith, M.; Viswanathan, M.
Corporate Source: IBM T.J. Watson Research Center, Yorktown Heights, NY 10598, United States
Conference Title: 2003 IEEE International Conference on Accoustics, Speech, and Signal Processing
Conference Location: Hong Kong, Hong Kong **Conference Date:** 20030406-20030410
E.I. Conference No.: 61464
Source: ICASSP, IEEE International Conference on Acoustics, Speech and Signal Processing - Proceedings v 1 2003. p 708-711 (IEEE cat n 03CH37404)
Publication Year: 2003
CODEN: IPRODJ **ISSN:** 0736-7791
Language: English
Author: Eide, E.; Aaron, A.; **Bakis, R.**; Cohen, P.; Donovan, R.; Hamza, W.; Mathes, T.; Picheny, M.; Polkosky, M.; Smith, M...
Abstract: ...led to significant gains in the output quality. On the algorithms side, we have introduced **statistical** models for **predicting pitch** and duration targets which replace the rule-based target generation previously employed. Additionally, we have...

19/3,K/4 (Item 1 from file: 144) **Links**
Pascal

15000077 PASCAL No.: 01-0155516

Methods for generating **pitch** and duration **contours** in a
text to speech system

EIDE Ellen M; DONOVAN Robert E
Journal: The Journal of the Acoustical Society of America
, 2001-04, 109 (4
) p. 1285
Language: English

Methods for generating **pitch** and duration **contours** in a
text to speech system
EIDE Ellen M; DONOVAN Robert E

[File 344] **Chinese Patents Abs** Jan 1985-2006/Jan

[File 347] **JAPIO** Dec 1976-2006/Dec(Updated 070403)

[File 350] **Derwent WPIX** 1963-2006/UD=200722

**File 350: DWPI has been enhanced to extend content and functionality of the database. For more info, visit http://www.dialog.com/dwpi/.*

[File 371] **French Patents** 1961-2002/BOPI 200209

**File 371: This file is not currently updating. The last update is 200209.*

```
Set      Items    Description
S1      216855    S (VOICE OR SPEECH OR SOUND OR NATURAL()LANGUAGE?? OR WORD OR
TEXT OR LEXIC? OR PHRASE? OR TEXT()STRING??)(3N)(SYNTHESI? OR GENERAT??? OR
RECOGNI? OR PROCESS? OR SEGMENT?? OR PORTION?? OR PART?? OR COMPONENT?? OR
SPETRA OR SPECTRUM OR BAND?? OR CONCAT?)
S2         530    S PITCH(3N)(PREDICT??? OR CONTOUR?? OR INTONATE??? OR
MODULAT???)
S3          59    S (CALCULAT? OR ESTIMAT??? OR DETERMIN? OR PROBABILIT??? OR
RANDOM OR STATISTIC??? OR STOCHASTIC)(3N)S2
S4         390    S (LOW()FREQUENC???)(3N)(ENERGY OR ENERGIES OR ENTROP?)
S5          14    S (INCREAS??? OR ADD?? OR IMPROV??? OR OPTIMI?)(3N)S4
S6           5    S FILTER??(3N)(S4 OR S5)
S7          38    S AU=(EIDE, E? OR EIDE E? OR BAKIS, R? OR BAKIS R?)
S8          13    S S5 NOT S6
S9           4    S S8 AND IC=G10L?
S10          1    S (S2 OR S3)(3N)(S4:S6)
S11          1    S (S2 OR S3)(S)(S4:S6)
S12          0    S S11 NOT S10
S13          2    S (S2 OR S3) AND (S4:S6)
S14          1    S S13 NOT S11
S15          4    S (S2 OR S3) AND S7
```

6/3,K/1 (Item 1 from file: 350) **Links**
Derwent WPIX

0014128411 *Drawing available*
WPI Acc no: 2004-312995/200429
XRPX Acc No: N2004-249122
Low-frequency filter for short-wave transmitter
Patent Assignee: OMSK TOOL CONSTR RES INST (OMTO-R)
Inventor: ALEKSEENKO V N; MINGALIEV T R

Patent Family (1 patents, 1 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
RU 2225672	C2	20040310	RU 2001131610	A	20011121	200429	B

Priority Applications (no., kind, date): RU 2001131610 A 20011121

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes
RU 2225672	C2	RU	0	1	

...capacitor 16, to output of inductance coil 3 adjacent to inductance coil 2. Part of **energy** in **low-frequency filter** where inductance coils 1 through 3 are not separated by screens is transferred from inductance...

6/3,K/2 (Item 2 from file: 350) **Links**
Derwent WPIX

0010154183 *Drawing available*
WPI Acc no: 2000-462981/200040
XRAM Acc no: C2000-139430
XRPX Acc No: N2000-345266
Voltage converter
Patent Assignee: INFOT RES PRODN ENTERPRISE CO LTD (INFO-R)
Inventor: MIKHAILOV B A

Patent Family (1 patents, 1 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
RU 2139623	C1	19991010	RU 1998122240	A	19981211	200040	B

Priority Applications (no., kind, date): RU 1998122240 A 19981211

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes
RU 2139623	C1	RU	0	1	

...switches; some design versions have also selector switch, input and output inductance units, diode, additional **energy** storage capacitors, **low-frequency filter**, rectifier, and high-voltage pulse shaper. Converter incorporates provision for reducing output voltage ripples and...

6/3,K/3 (Item 3 from file: 350) **Links**
Derwent WPIX

0004404100 *Drawing available*
WPI Acc no: 1988-140694/198821
Miniature liner prediction vocoder with signal analysis appts. - hamming windows digital signal and determines number of auto-correlation coefficients including zero lag representing residual energy
Patent Assignee: CANAD MIN MIN COMMU (CNDG)
Inventor: BRYDEN B; HASSANEIN H R

Patent Family (1 patents, 1 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
CA 1235815	A	19880426	CA 656657	A	19841001	198821	B
			CA 488381	A	19850808		

Priority Applications (no., kind, date): US 1984656657 A 19841001

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes
CA 1235815	A	EN	74	17	

Alerting Abstract ...partial reflection coefficients are calculated from the autocorrelation coefficients. The digital signal is low pass **filtered** and a **low frequency energy** value is determined as is the rate of zero crossings of the input

signal...

6/3,K/4 (Item 4 from file: 350) **Links**
Derwent WPIX

0002812115
WPI Acc no: 1983-B3730K/198305
Data MODEM clock extraction circuit for envelope recovery - operating on squared differences between delayed and undelayed samples to form PLL corrected at sampling frequency
Patent Assignee: RACAL DATA COMMUNICATIONS INC (RACA)
Inventor: KROMER P F; KRONER P F

Patent Family (13 patents, 16 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
BE 894459	A	19830117				198305	B
WO 1983001165	A	19830331	WO 1982US1261	A	19820917	198314	E
FR 2513460	A	19830325				198317	E
AU 198289584	A	19830408				198326	E
JP 58501491	W	19830901				198341	E
DE 3249021	T	19831117				198347	E
GB 2122850	A	19840118	GB 198312019	A	19820923	198403	E
US 4455665	A	19840619	US 1981304044	A	19810921	198427	E
GB 2122850	B	19850731	GB 198212019	A	19820917	198531	E
CA 1196992	A	19851119				198551	E
KR 198801166	A	19880702				198845	E
DE 3249021	C	19910425	DE 3249021	A	19820917	199117	E
IT 1212667	B	19891130				199150	E

Priority Applications (no., kind, date): US 1981304044 A 19810921

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes	
BE 894459	A	FR	12			
WO 1983001165	A	EN				
National Designated States,Original	AU BR CH DE DK GB JP NL NO SE					
CA 1196992	A	EN				

Original Publication Data by Authority

6/3,K/5 (Item 5 from file: 350) **Links**
Derwent WPIX

0001789962
WPI Acc no: 1979-D3763B/197915
Defibrillator monitor baseline control - has filter connected to ECG electrodes applied to patient's body, for increasing-attenuation of LF energy passing through amplifier
Patent Assignee: HEWLETT-PACKARD CO (HEWP)
Inventor: GATZKE R D

Patent Family (1 patents, 1 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
US 4147162	A	19790403	US 1977805550	A	19770610	197915	B

Priority Applications (no., kind, date): US 1977805550 A 19770610

Alerting Abstract ...which is derived from the electrodes prior to its application to the input of the **filter** for **increasing** the attenuation of **low frequency energy.**

9/3,K/1 (Item 1 from file: 350) **Links**
Derwent WPIX

0012823657 *Drawing available*
WPI Acc no: 2002-681353/200273
XRPX Acc No: N2002-537798
Speech synthesizing method for speech-to-speech translation, involves increasing energy in low frequency components of pitch contour generated for synthesized speech
Patent Assignee: BAKIS R (BAKI-I); EIDE E M (EIDE-I)
Inventor: BAKIS R; EIDE E M

Patent Family (1 patents, 1 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
US 20020072909	A1	20020613	US 2000732122	A	20001207	200273	B

Priority Applications (no., kind, date): US 2000732122 A 20001207

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes
US 20020072909	A1	EN	8	4	

Speech synthesizing method for speech-to-speech translation, involves increasing energy in low frequency components of pitch contour generated for synthesized speech Class Codes International Patent Classification IPC Class Level Scope Position Status Version Date **G10L-013/06** Main **G10L-013/00** Original Publication Data by Authority...**Claims:**speech; and increasing an amount of energy in low frequency components of said pitch contour.

9/3,K/2 (Item 2 from file: 350) **Links**
Derwent WPIX

0010967483 *Drawing available*
WPI Acc no: 2001-591108/200167
Related WPI Acc No: 1994-359896; 2001-531148
XRPX Acc No: N2001-440358
Speech detection apparatus for detecting speech period, in a video conference system, and an audio reproduction system of television or audio equipment
Patent Assignee: MATSUSHITA ELECTRIC IND CO LTD (MATU)
Inventor: NAKATOH Y; NORIMATSU T

Patent Family (3 patents, 2 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
EP 1083542	A2	20010314	EP 1994107786	A	19940519	200167	B
			EP 2000123382	A	19940519		
EP 1083542	B1	20031015	EP 1994123382	A	19940519	200368	E
DE 69433254	E	20031120	DE 69433254	A	19940519	200401	E
			EP 2000123382	A	19940519		

Priority Applications (no., kind, date): EP 1994123382 A 19940519; JP 1993116980 A 19930519

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes	
EP 1083542	A2	EN	48	24	Division of application	EP 1994107786
					Division of patent	EP 625774

Regional Designated States,Original	DE FR GB					
EP 1083542	B1	EN				
Regional Designated States,Original	DE FR GB					
DE 69433254	E	DE			Application	EP 2000123382
					Based on OPI patent	EP 1083542

Alerting Abstract ...performing speech detection and preventing an erroneous decision, even if stationary noises or noises whose **energy** predominates in the **low-frequency** region are **added** to the speech... **Class Codes** International Patent Classification IPC Class Level Scope Position Status Version Date **G10L-011/02** Main

9/3,K/3 (Item 3 from file: 350) **Links**
Derwent WPIX

0008104441 *Drawing available*
WPI Acc no: 1997-202424/199718
XRPX Acc No: N1997-167276
Adaptive filtering of audio signals to enhance speech intelligibility - reduces undesirable encoded background noise while minimising effect on speech quality and digital processing resource power consumption
Patent Assignee: ERICSSON INC (TELF)
Inventor: SOELVE T W; SOLVE T W

Patent Family (15 patents, 70 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
WO 1997010586	A1	19970320	WO 1996US14665	A	19960913	199718	B
AU 199670784	A	19970401	AU 199670784	A	19960913	199730	E
NO 199801074	A	19980513	WO 1996US14665	A	19960913	199829	E
			NO 19981074	A	19980311		
EP 852052	A1	19980708	EP 1996931552	A	19960913	199831	E
			WO 1996US14665	A	19960913		
CN 1201547	A	19981209	CN 1996198008	A	19960913	199917	E
BR 199610290	A	19990316	BR 199610290	A	19960913	199918	E
			WO 1996US14665	A	19960913		
JP 11514453	W	19991207	WO 1996US14665	A	19960913	200008	E
			JP 1997512112	A	19960913		
MX 199801857	A1	19981101	MX 19981857	A	19980309	200022	E
KR 1999044659	A	19990625	WO 1996US14665	A	19960913	200036	E
			KR 1998701913	A	19980313		
AU 724111	B	20000914	AU 199670784	A	19960913	200051	E
RU 2163032	C2	20010210	WO 1996US14665	A	19960913	200122	E
			RU 1998107313	A	19960913		
EP 852052	B1	20010613	EP 1996931552	A	19960913	200134	E
			WO 1996US14665	A	19960913		
DE 69613380	E	20010719	DE 69613380	A	19960913	200148	E
			EP 1996931552	A	19960913		
			WO 1996US14665	A	19960913		
KR 423029	B	20040701	WO 1996US14665	A	19960913	200471	E
			KR 1998701913	A	19980313		
CN 1121684	C	20030917	CN 1996198008	A	19960913	200552	E

Priority Applications (no., kind, date): WO 1996US14665 A 19960913; US 1995528005 A 19950914

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes

WO 1997010586	A1	EN	54	12		
National Designated States,Original	AL AM AT AU AZ BA BB BG BR BY CA CH CN CU CZ DE DK EE ES FI GB GE HU IL IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK TJ TM TR TT UA UG UZ VN					
Regional Designated States,Original	AT BE CH DE DK EA ES FI FR GB GR IE IT KE LS LU MC MW NL OA PT SD SE SZ UG					
AU 199670784	A	EN			Based on OPI patent	WO 1997010586
NO 199801074	A	NO			PCT Application	WO 1996US14665
EP 852052	A1	EN			PCT Application	WO 1996US14665
					Based on OPI patent	WO 1997010586
Regional Designated States,Original	BE DE DK ES FI FR GB GR IT NL PT SE					
BR 199610290	A	PT			PCT Application	WO 1996US14665
					Based on OPI patent	WO 1997010586
JP 11514453	W	JA	51		PCT Application	WO 1996US14665
					Based on OPI patent	WO 1997010586
KR 1999044659	A	KO		16	PCT Application	WO 1996US14665
					Based on OPI patent	WO 1997010586
AU 724111	B	EN			Previously issued patent	AU 9670784
					Based on OPI patent	WO 1997010586
RU 2163032	C2	RU			PCT Application	WO 1996US14665
					Based on OPI patent	WO 1997010586
EP 852052	B1	EN			PCT Application	WO 1996US14665
					Based on OPI patent	WO 1997010586
Regional Designated States,Original	BE DE DK ES FI FR GB GR IT NL PT SE					
DE 69613380	E	DE			Application	EP 1996931552
					PCT Application	WO 1996US14665
					Based on OPI patent	EP 852052
					Based on OPI patent	WO 1997010586
KR 423029	B	KO			PCT Application	WO 1996US14665
					Previously issued patent	KR 99044659
					Based on OPI patent	WO 1997010586

Class Codes International Patent Classification IPC Class Level Scope Position Status Version Date **G10L**; ... ...**G10L-019/08**... ...**G10L-021/00**... ...**G10L-021/02**... ...**G10L-003/02**... ...**G10L-009/00** Main **G10L-015/20** Original Publication Data by Authority...**Original Abstracts:**the noise estimates increase, the filter circuit is adjusted to extract increasing amounts of energy **falling** in low **frequency** ranges of **speech**. **In** a second preferred embodiment, the filter circuit is adjusted as a function of a noise... ... filter circuit is adjusted to extract increasing amounts of energy falling in low frequency ranges **of** speech. In **a** second preferred **embodiment**, **the** filter circuit is adjusted as a function of a noise profile estimate. A noise profile...

9/3,K/4 (Item 4 from file: 350) **Links**
Derwent WPIX

0006958204 *Drawing available*
WPI Acc no: 1994-359896/199445
Related WPI Acc No: 2001-531148; 2001-591108
XRPX Acc No: N1994-281998
Speech detection appts. for video conference or speech recognition etc. - compares parameters extracted from each frame with reference model to decide if audio signal is speech or not
Patent Assignee: MATSUSHITA ELEC IND CO LTD (MATU); MATSUSHITA ELECTRIC IND CO LTD (MATU)
Inventor: NAKATO Y; NAKATOH Y; NORIMATSU T

Patent Family (5 patents, 2 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
EP 625774	A2	19941123	EP 1994107786	A	19940519	199445	B

EP 625774	A3	19961030	EP 1994107786	A	19940519	199649	E
US 5611019	A	19970311	US 1994246346	A	19940519	199716	E
EP 625774	B1	20020313	EP 1994107786	A	19940519	200219	E
			EP 2000123381	A	19940519		
			EP 2000123382	A	19940519		
DE 69430082	E	20020418	DE 69430082	A	19940519	200234	E
			EP 1994107786	A	19940519		

Priority Applications (no., kind, date): JP 1993116980 A 19930519

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes	
EP 625774	A2	EN	56	24		
Regional Designated States,Original	DE FR GB					
EP 625774	A3	EN				
US 5611019	A	EN	44	24		
EP 625774	B1	EN			Related to application	EP 2000123381
					Related to application	EP 2000123382
					Related to patent	EP 1083541
					Related to patent	EP 1083542
Regional Designated States,Original	DE FR GB					
DE 69430082	E	DE			Application	EP 1994107786
					Based on OPI patent	EP 625774

Alerting Abstract ...noisy environment. Accurate speech detection using simple appts., even if stationary noises or noises whose **energy** predominates in the **low-frequency** region are **added** to the speech. **Class Codes** International Patent Classification IPC Class Level Scope Position Status Version Date **G10L-011/02**... ...**G10L-003/00**... ...**G10L-005/06** Main

10/3,K/1 (Item 1 from file: 350) Links
Derwent WPIX

0012823657 *Drawing available*
WPI Acc no: 2002-681353/200273
XRPX Acc No: N2002-537798
Speech synthesizing method for speech-to-speech translation, involves increasing energy in low frequency components of pitch contour generated for synthesized speech
Patent Assignee: BAKIS R (BAKI-I); EIDE E M (EIDE-I)
Inventor: BAKIS R; EIDE E M

Patent Family (1 patents, 1 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
US 20020072909	A1	20020613	US 2000732122	A	20001207	200273	B

Priority Applications (no., kind, date): US 2000732122 A 20001207

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes
US 20020072909	A1	EN	8	4	

Speech synthesizing method for speech-to-speech translation, involves increasing energy in low frequency components of pitch contour generated for synthesized speech Alerting Abstract ...NOVELTY - A pitch contour

for synthesized speech is generated. The amount of **energy** in the **low frequency** components of the **pitch contour**, is increased. ...ADVANTAGE - The naturalness of sounding speech is achieved, as the **energy** in **low frequency** components of **pitch contour** is increased... Original Publication Data by Authority...**Claims:**speech; and increasing an amount of energy in low frequency components of said pitch contour.

14/3,K/1 (Item 1 from file: 350) **Links**
Derwent WPIX

0010393004 *Drawing available*
WPI Acc no: 2000-387053/200033
Related WPI Acc No: 2000-411373; 2000-411376; 2000-411377
XRPX Acc No: N2000-289789
Periodicity enhancing device of excitation signal, reduces energy of low frequency portion of innovative code vector in relation to periodicity factor related to wideband signal
Patent Assignee: UNIV SHERBROOKE (UYSH); VOICEAGE CORP (VOIC-N)
Inventor: BESSETTE B; LEFEBRE R; LEFEBVRE R; SALAMI R

Patent Family (20 patents, 89 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
WO 2000025303	A1	20000504	WO 1999CA1009	A	19991027	200033	B
CA 2252170	A1	20000427	CA 2252170	A	19981027	200038	E
AU 199964570	A	20000515	AU 199964570	A	19991027	200039	E
EP 1125285	A1	20010822	EP 1999952200	A	19991027	200149	E
			WO 1999CA1009	A	19991027		
CN 1328682	A	20011226	CN 1999813602	A	19991027	200227	E
ZA 200103366	A	20020731	ZA 20013366	A	20010425	200271	E
ZA 200103367	A	20020731	ZA 20013367	A	20010425	200271	E
JP 2002528983	W	20020903	WO 1999CA1009	A	19991027	200273	E
			JP 2000578810	A	19991027		
EP 1125285	B1	20030730	EP 1999952200	A	19991027	200356	E
			WO 1999CA1009	A	19991027		
DE 69910058	E	20030904	DE 69910058	A	19991027	200366	E
			EP 1999952200	A	19991027		
			WO 1999CA1009	A	19991027		
ES 2205892	T3	20040501	EP 1999952200	A	19991027	200431	E
US 6795805	B1	20040921	WO 1999CA1009	A	19991027	200462	E
			US 2001830331	A	20010723		
US 20050108005	A1	20050519	WO 1999CA1008	A	19991027	200534	E
			US 2001830114	A	20010620		
			US 2004964752	A	20041015		
US 20050108007	A1	20050519	WO 1999CA1010	A	19991027	200534	E
			US 2001830276	A	20010620		
			US 2004965795	A	20041018		
CN 1127055	C	20031105	CN 1999813602	A	19991027	200556	E
CA 2347667	C	20060214	CA 2347667	A	19991027	200615	E
			WO 1999CA1009	A	19991027		
CN 1165891	C	20040908	CN 1999813640	A	19991027	200615	E
CN 1165892	C	20040908	CN 1999813641	A	19991027	200615	E
IN 200100435	P2	20060310	WO 1999CA1010	A	19991027	200626	E
			IN 2001KN435	A	20010418		
JP 3869211	B2	20070117	WO 1999CA1009	A	19991027	200707	E
			JP 2000578810	A	19991027		

Priority Applications (no., kind, date): CA 2252170 A 19981027

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes	
WO 2000025303	A1	EN	81	4		

National Designated States,Original	AE AL AM AT AU AZ BA BB BG BR BY CA CH CN CR CU CZ DE DK DM EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MA MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT TZ UA UG US UZ VN YU ZA ZW					
Regional Designated States,Original	AT BE CH CY DE DK EA ES FI FR GB GH GM GR IE IT KE LS LU MC MW NL OA PT SD SE SL SZ TZ UG ZW					
CA 2252170	A1	EN				
AU 199964570	A	EN			Based on OPI patent	WO 2000025303
EP 1125285	A1	EN			PCT Application	WO 1999CA1009
					Based on OPI patent	WO 2000025303
Regional Designated States,Original	AT BE CH CY DE DK ES FI FR GB GR IE IT LI LU MC NL PT SE					
ZA 200103366	A	EN	72			
ZA 200103367	A	EN	76			
JP 2002528983	W	JA	62		PCT Application	WO 1999CA1009
					Based on OPI patent	WO 2000025303
EP 1125285	B1	EN			PCT Application	WO 1999CA1009
					Based on OPI patent	WO 2000025303
Regional Designated States,Original	AT BE CH CY DE DK ES FI FR GB GR IE IT LI LU MC NL PT SE					
DE 69910058	E	DE			Application	EP 1999952200
					PCT Application	WO 1999CA1009
					Based on OPI patent	EP 1125285
					Based on OPI patent	WO 2000025303
ES 2205892	T3	ES			Application	EP 1999952200
					Based on OPI patent	EP 1125285
US 6795805	B1	EN			PCT Application	WO 1999CA1009
					Based on OPI patent	WO 2000025303
US 20050108005	A1	EN			Division of application	WO 1999CA1008
					Division of application	US 2001830114
US 20050108007	A1	EN			Continuation of application	WO 1999CA1010
					Continuation of application	US 2001830276
					Continuation of patent	US 6807524
CA 2347667	C	EN			PCT Application	WO 1999CA1009
					Based on OPI patent	WO 2000025303
IN 200100435	P2	EN			PCT Application	WO 1999CA1010
JP 3869211	B2	JA	36		PCT Application	WO 1999CA1009
					Previously issued patent	JP 2002528983
					Based on OPI patent	WO 2000025303

Periodicity enhancing device of excitation signal, reduces energy of low frequency portion of innovative code vector in relation to periodicity factor related to wideband signal ...filters the innovative code vector in relation to the periodicity factor, so as to reduce **energy** of **low frequency** portion of the innovative code vector and enhance periodicity of low frequency portion of the... Original Publication Data by Authority...**Original Abstracts:**innovation filter subsequently processes the innovative codevector in relation to this periodicity factor to reduce **energy** of a **low frequency** portion of the innovative codevector and enhance periodicity of a low frequency portion of the... ... filters applied to a pitch codevector, the one yielding higher prediction gain (i.e. the **lowest pitch** prediction error) is selected and the associated pitch codebook parameters are forwarded... ... An innovation filter subsequently processes the innovative codevector in relation to this periodicity factor to **reduce** energy of **a low** frequency portion of the innovative codevector and enhance periodicity of a low frequency portion of... ...**Claims:**205) for filtering the innovative codevector in relation to said periodicity factor to thereby reduce **energy** of a **low frequency** portion of the innovative codevector and enhance periodicity of a low frequency portion of the... ... associated to respective sets of pitch codebook parameters, wherein:i) each signal path comprises a **pitch prediction** error **calculating** device for **calculating** a **pitch prediction** error of a **pitch** codevector from a pitch codebook search device; andii) at least one of said two... ... comprises a filter for filtering the pitch codevector before supplying said pitch codevector to the **pitch prediction** error **calculating** device of said one path; andb) a selector for comparing the **pitch prediction** errors **calculated** in said at least two signal paths, for choosing the signal path having the lowest **calculated pitch prediction** error, and for selecting the set of pitch codebook parameters associated to the choosen signal... ... filter for filtering the

innovative codevector in relation to said periodicity factor to thereby reduce **energy** of a **low frequency** portion of the innovative codevector and enhance periodicity of a low frequency portion of the...

15/3,K/1 (Item 1 from file: 350) **Links**
Derwent WPIX

0016383768 *Drawing available*
WPI Acc no: 2007-099941/200710
XRPX Acc No: N2007-069935
Speech synthesizing method for human listener, involves determining smooth pitch contour between adjacent anchor points by linearly interpolating between pitch values of smooth pitch contour at anchor points
Patent Assignee: BAKIS R (BAKI-I)
Inventor: **BAKIS R**

Patent Family (1 patents, 1 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
US 20060259303	A1	20061116	US 2005128003	A	20050512	200710	B

Priority Applications (no., kind, date): US 2005128003 A 20050512

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes
US 20060259303	A1	EN	14	6	

Speech synthesizing method for human listener, involves determining smooth pitch contour between adjacent anchor points by linearly interpolating between pitch values of smooth pitch contour at anchor points Inventor: **BAKIS R Alerting Abstract** ...The method involves generating a sequence of phonetic units representative of a target utterance, and **determining** a **pitch contour** for the target utterance, where the **pitch contour** has a set of linear **pitch contour** segments. The **pitch contour** is filtered to **determine** pitch values of a smooth **pitch contour** at anchor points. The smooth **pitch contour** is **determined** between adjacent anchor points by linearly interpolating between the pitch values of the smooth **pitch contour** at the anchor points. ... ADVANTAGE - The method enables fast and efficient smoothing of discontinuous, non-smooth **pitch contours** obtained from concatenation of the speech segments while improving a quality of a synthesized signal... Original Publication Data by AuthorityInventor name & address:**Bakis, Raimo**... **Original Abstracts:**TTS synthesis systems are provided which implement computationally fast and efficient **pitch contour** smoothing methods for determining smooth **pitch contours** for non-smooth **pitch contours**, which closely track the non-smooth **pitch contours**. For example, a TTS method includes generating a sequence of phonetic units representative of a target utterance, **determining** a **pitch contour** for the target utterance, the **pitch contour** comprising a plurality of linear **pitch contour** segments, wherein each linear **pitch contour** segment has start and end times at anchor points of the **pitch contour**, filtering the **pitch contour** to **determine pitch** values of a smooth **pitch contour** at the anchor points, and **determining** the smooth **pitch contour** between adjacent anchor points by linearly interpolating between the pitch values of the smooth **pitch contour** at the anchor points. ...**Claims:**for speech synthesis, comprising:generating a sequence of phonetic units representative of a target utterance;**determining** a **pitch contour** for the target utterance, the **pitch contour** comprising a plurality of linear **pitch contour** segments, wherein each linear **pitch contour** segment has start and end times at anchor points of the **pitch contour**;filtering the **pitch contour** to **determine pitch** values of a smooth **pitch contour** at the anchor points; and**determining** the smooth **pitch contour** between adjacent anchor points by linearly interpolating between the pitch values of the smooth **pitch contour** at the anchor points.

15/3,K/2 (Item 2 from file: 350) **Links**
Derwent WPIX

0014982537 *Drawing available*
WPI Acc no: 2005-330386/200534
XRPX Acc No: N2005-270140
Program storage device in text-to-speech conversion system, stores instructions for automatically generating marked-up text corresponding to spoken utterance using prosodic parameters such as pitch contour and

duration contour of utterance
Patent Assignee: INT BUSINESS MACHINES CORP (IBMC)
Inventor: AARON A; **BAKIS R**; **EIDE E M**; HAMZA W M

Patent Family (1 patents, 1 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
US 20050071163	A1	20050331	US 2003672374	A	20030926	200534	B

Priority Applications (no., kind, date): US 2003672374 A 20030926

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes
US 20050071163	A1	EN	9	4	

...**for automatically generating marked-up text corresponding to spoken utterance using prosodic parameters such as pitch contour and duration contour of utterance** ...Inventor: **BAKIS R**... ...**EIDE E M** ...NOVELTY - The prosodic parameters such as **pitch contour**, duration **contour**, energy contour information of spoken utterance are determined. A marked-up text corresponding to the... Original Publication Data by Authority...Inventor name & address:**Bakis, Raimo**... ...**Eide, Ellen M**

15/3,K/3 (Item 3 from file: 350) Links
Derwent WPIX

0012823657 *Drawing available*
WPI Acc no: 2002-681353/200273
XRPX Acc No: N2002-537798
Speech synthesizing method for speech-to-speech translation, involves increasing energy in low frequency components of pitch contour generated for synthesized speech
Patent Assignee: BAKIS R (BAKI-I); EIDE E M (EIDE-I)
Inventor: **BAKIS R**; **EIDE E M**

Patent Family (1 patents, 1 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
US 20020072909	A1	20020613	US 2000732122	A	20001207	200273	B

Priority Applications (no., kind, date): US 2000732122 A 20001207

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes
US 20020072909	A1	EN	8	4	

...**synthesizing method for speech-to-speech translation, involves increasing energy in low frequency components of pitch contour generated for synthesized speech Original Titles:**Method and apparatus for producing natural sounding **pitch contours** in a speech synthesizer Inventor: **BAKIS R**... ...**EIDE E M Alerting Abstract** ...NOVELTY - A **pitch contour** for synthesized speech is generated. The amount of energy in the low frequency components of the **pitch contour**, is increased. ...The naturalness of sounding speech is achieved, as the energy in low frequency components of **pitch contour** is increased... Original Publication Data by AuthorityInventor name & address:**Eide, Ellen Marie**... ...**Bakis, Raimo Original Abstracts:** A speech synthesis system is disclosed that utilizes a pitch contour resulting in **a more** natural-sounding speech. The present invention modifies the predicted pitch, b(t), **for synthesized** speech using a low frequency energy booster. The low frequency energy booster interpolates the discrete... ... values, if necessary, and increase the amount of energy of the pitch contour associated with **low frequency** values, such as all frequency values below 10 Hertz. The amount of energy of the pitch contour associated with **low frequency** values can be increased, for example, by adding band-limited noise (a carrier signal) to the pitch contour, b(t), **or by** filtering the pitch values with an impulse response filter having a pole at the desired... ... present invention serves to add

vibrato to the to the original pitch contour, b(t), **and thereby** improves the naturalness of the synthetic waveform. ...**Claims:**speech, comprising: generating a pitch contour for said synthesized speech; and increasing an amount of **energy in** low frequency components of said pitch contour.

15/3,K/4 (Item 4 from file: 350) **Links**
Derwent WPIX

0010273264 *Drawing available*
WPI Acc no: 2000-586104/200055
XRPX Acc No: N2000-433676
Pitch contours generation in text-to-speech system, involves determining stored stress level closer to stress levels of input text and copying pitch levels associated with those of stress and pitch level pairs
Patent Assignee: INT BUSINESS MACHINES CORP (IBMC)
Inventor: DONOVAN R E; **EIDE E M**

Patent Family (1 patents, 1 countries)

Patent Number	Kind	Date	Application Number	Kind	Date	Update	Type
US 6101470	A	20000808	US 199884679	A	19980526	200055	B

Priority Applications (no., kind, date): US 199884679 A 19980526

Patent Details

Patent Number	Kind	Lan	Pgs	Draw	Filing Notes
US 6101470	A	EN	15	6	

Pitch contours generation in text-to-speech system, involves determining stored stress level closer to stress levels... Original Titles: Methods for generating **pitch** and duration **contours** in a text to speech system. ...Inventor: **EIDE E M Alerting Abstract** ...associated with closest stress levels of stress and pitch level pairs are copied to generate **pitch contours** of input text. ...stress level corresponding to secondary stress, and second stress level corresponding to primary stress. A **pitch contour** model is trained based on training text read by at least one speaker to generate... ...training sentences. The sequences of stress and pitch level pairs correspond to training sentences. The **pitch contour** of training sentences is calculated from laryngograph data as a function of time by noting... ...USE - For generating **pitch contours** in text-to-speech (TtS) system... ...quality of synthesized speech improves with the number of training utterances available for selecting the **pitch contour** to be synthesized, the use of only lexical stress contours as features for selecting the **pitch contour** enables a relatively small, efficiently searched database of **pitch contours** to suffice for very good quality prosody in synthesis. A smaller number of sentences are... Original Publication Data by AuthorityInventor name & address:**Eide, Ellen M**... **Original Abstracts:**A method for automatically generating **pitch contours** in a text to speech (TtS) system, the system converting input text into an output... ...the closest stored stress levels of the stress and pitch level pairs to generate the **pitch contours** of the input text. Features illustrative of various modes of the invention include stress and... **Claims:**A method for generating pitch contours **in a** text to speech (TtS) system, the system converting input text into an output acoustic signal

[File 348] **EUROPEAN PATENTS** 1978-2007/ 200713

**File 348: For important information about IPCR/8 and forthcoming changes to the IC= index, see HELP NEWSIPCR.*

[File 349] **PCT FULLTEXT** 1979-2007/UB=20070329UT=20070322

**File 349: For important information about IPCR/8 and forthcoming changes to the IC= index, see HELP NEWSIPCR.*

```
Set     Items   Description
S1      91506   S (VOICE OR SPEECH OR SOUND OR NATURAL()LANGUAGE?? OR WORD OR
TEXT OR LEXIC? OR PHRASE? OR TEXT()STRING??)(3N)(SYNTHESI? OR GENERAT??? OR
RECOGNI? OR PROCESS? OR SEGMENT?? OR PORTION?? OR PART?? OR COMPONENT?? OR
SPETRA OR SPECTRUM OR BAND?? OR CONCAT?)
S2       1195   S PITCH(3N)(PREDICT??? OR CONTOUR?? OR INTONATE??? OR
MODULAT???)
S3        203   S (CALCULAT? OR ESTIMAT??? OR DETERMIN? OR PROBABILIT??? OR
RANDOM OR STATISTIC??? OR STOCHASTIC)(3N)S2
S4       1084   S (LOW()FREQUENC???)(3N)(ENERGY OR ENERGIES OR ENTROP?)
S5         42   S (INCREAS??? OR ADD?? OR IMPROV??? OR OPTIMI?)(3N)S4
S6         29   S FILTER??(3N)(S4 OR S5)
S7          5   S AU=(EIDE, E? OR EIDE E? OR BAKIS, R? OR BAKIS R?)
S8          0   S (S2 OR S3)(40N)(S4:S6)
S9         20   S (S2 OR S3) AND (S4:S6)
S10        18   S S9 AND IC=G10L?
S11        11   S S10 AND PITCH()CONTOUR??
S12        18   S S10 AND FILTER???
S13         7   S S12 NOT S11
S14       132   S (S2 OR S3)(3N)FILTER???
S15         0   S S14(3N)S4
S16         0   S S14(S)S4
S17         4   S S14 AND S4
S18         0   S S17 NOT (S11 OR S13)
S19         5   S S7 NOT (S11 OR S13 OR S17)
S20         2   S S19 AND IC=G10L?
```

11/3K/1 (Item 1 from file: 348) Links
EUROPEAN PATENTS

01275156
TELEPHONIC EMOTION DETECTOR WITH OPERATOR FEEDBACK
TELEFONISCHER EMOTIONSDETEKTOR MIT RUCKMELDUNG AN einen BEDIENER
DETECTEUR D'EMOTIONS TELEPHONIQUE AVEC RETOUR A un OPERATEUR

Patent Assignee:

- **Accenture LLP**; (3330220)
  1661 Page Mill Road; Palo Alto, CA 94304; (US)
  (Proprietor designated states: all)

Inventor:

- **PETRUSHIN, Valery, A.**
  871 Stonebridge Lane; Buffalo Grove IL 60089; (US)

Legal Representative:

- **McLeish, Nicholas Alistair Maxwell (74621)**
  Boult Wade Tennant Verulam Gardens 70 Gray's Inn Road; London WC1X 8BT; (GB)

	Country	Number	Kind	Date	
Patent	EP	1222656	A1	20020717	(Basic)
	EP	1222656	B1	20050615	
	WO	2001016939		20010308	
Application	EP	2000961546		20000831	
	WO	2000US24325		20000831	
Priorities	US	387621		19990831	

Designated States:
AT; BE; CH; CY; DE; DK; ES; FI; FR; GB;
GR; IE; IT; LI; LU; MC; NL; PT;

Extended Designated States:
AL; LT; LV; MK; RO; SI;

International Patent Class (V7): G10L-017/00; G10L-017/00
NOTE: No A-document published by EPO

Type	Pub. Date	Kind	Text
Publication: English			
Procedural: English			
Application: English			

Available Text	Language	Update	Word Count
CLAIMS B	(English)	200524	919
CLAIMS B	(German)	200524	946
CLAIMS B	(French)	200524	951
SPEC B	(English)	200524	18452
Total Word Count (Document A) 0			
Total Word Count (Document B) 21268			
Total Word Count (All Documents) 21268			

Specification: ...features such as LPC (linear predictive coding) parameters of signal or features of the smoothed **pitch contour** and its derivatives.

For this invention, the following strategy may be adopted. First, take into... ...a linear regression for voiced part of speech, i.e. the line that fits the **pitch contour**. The relative voiced energy can also be calculated as the proportion of voiced energy to...in effect correspond to resonances of the vocal tract, the human voice also contains a **pitch**, **modulated** by the speaker, which corresponds to the frequency at which the larynx modulates the air... ...speech to text systems.

The first reflection coefficient k1)) is approximately related to the high/**low frequency energy** ratio and a signal. See R. J. McAulay, "Design of a Robust Maximum Likelihood Pitch... ...1979--28, Lincoln Labs, June 11, 1979. For k1)) close to -1, there is more **low frequency energy** in the signal than high-frequency energy, and vice versa for k1)) close to 1... ...and their goodness values C(k), dynamic programming is now used to obtain an optimum **pitch contour** which includes an optimum voicing decision for each frame. The dynamic programming requires several frames...

11/3K/2 (Item 2 from file: 348) Links
EUROPEAN PATENTS

01275093
DETECTING EMOTION IN VOICE SIGNALS THROUGH ANALYSIS OF A PLURALITY OF VOICE SIGNAL PARAMETERS
DETEKTION VON EMOTIONEN IN SPRACHSIGNALEN MITTELS ANALYSE EINER VIELZAHL VON SPRACHSIGNALPARAMETERN
DETECTION DES EMOTIONS DANS LES SIGNAUX VOCAUX PAR ANALYSE D'UNE PLURALITE DE PARAMETRES DE SIGNAUX VOCAUX

Patent Assignee:

- **Accenture LLP**; (3330220)
  1661 Page Mill Road; Palo Alto, CA 94304; (US)
  (Proprietor designated states: all)

Inventor:

- **PETRUSHIN, Valery, A.**
  871 Stonebridge Lane; Buffalo Grove, IL 60089; (US)

Legal Representative:

- **McLeish, Nicholas Alistair Maxwell et al (74621)**
  Boult Wade Tennant Verulam Gardens 70 Gray's Inn Road; London WC1X 8BT; (GB)

	Country	Number	Kind	Date	
Patent	EP	1125280	A1	20010822	(Basic)
	EP	1125280	B1	20070124	
	WO	2001016938		20010308	
Application	EP	2000961438		20000831	
	WO	2000US23884		20000831	
Priorities	US	388027		19990831	

Designated States:
AT; BE; CH; CY; DE; DK; ES; FI; FR; GB;
GR; IE; IT; LI; LU; MC; NL; PT; SE;

Extended Designated States:
AL; LT; LV; MK; RO; SI;

International Patent Class (V7): G10L-017/00; G10L-017/00

IPC	Level	Value	Position	Status	Version	Action	Source	Office
G10L-0017/00	A	I	F	B	20060101	20010315	H	EP
G10L-0017/00	A	I	F	B	20060101	20010315	H	EP

NOTE: No A-document published by EPO

Type	Pub. Date	Kind	Text
Publication: English			
Procedural: English			
Application: English			

Available Text	Language	Update	Word Count
CLAIMS B	(English)	200704	985
CLAIMS B	(German)	200704	906
CLAIMS B	(French)	200704	1068
SPEC B	(English)	200704	34707
Total Word Count (Document A) 0			
Total Word Count (Document B) 37666			
Total Word Count (All Documents) 37666			

Specification: ...features such as LPC (linear predictive coding) parameters of signal or features of the smoothed **pitch contour** and its derivatives.

The following strategy may be adopted. First, take into account fundamental frequency... ...a linear regression for voiced part of speech, i.e. the line that fits the **pitch contour**. The relative voiced energy can also be calculated as the proportion of voiced energy to...in effect correspond to resonances of the vocal tract, the human voice also contains a **pitch, modulated** by the speaker, which corresponds to the frequency at which the larynx modulates the air... ...speech to text systems.

The first reflection coefficient k1)) is approximately related to the high/**low frequency energy** ratio and a signal. See R. J. McAulay, "Design of a Robust Maximum Likelihood Pitch... ...1979, which is hereby incorporated by reference.

For k1)) close to -1, there is more **low frequency energy** in the signal than high-frequency energy, and vice versa for k1)) close to 1...and their goodness values C(k), dynamic programming is now used to obtain an optimum **pitch contour** which includes an optimum voicing decision for each frame. The dynamic programming requires several frames...

11/3K/3 (Item 3 from file: 348) **Links**
EUROPEAN PATENTS

01142718
ADAPTIVE TILT COMPENSATION FOR SYNTHESIZED SPEECH RESIDUAL
ADAPTIVE KOMPENSATION DER SPEKTRALEN VERZERRUNG EINES SYNTHETISIERTEN SPRACHRESIDUUMS
COMPENSATION D'INCLINAISONS ADAPTATIVE POUR RESIDUS VOCAUX SYNTHETISES

Patent Assignee:

- **Conexant Systems, Inc.**; (2732580)
  4311 Jamboree Road; Newport Beach, California 92660-3095; (US)
  (Applicant designated States: all)

Inventor:

- **YANG, Gao**
  26586 San Torini Road; Mission Viejo, CA 92692-6101; (US)
- **HUAN-YU, Su**
  3009 Calle Frontera; San Clemente, CA 92673-3029; (US)

Legal Representative:

- **Bayliss, Geoffrey Cyril (28154)**
  BOULT WADE TENNANT Verulam Gardens 70 Gray's Inn Road; London WC1X 8BT; (GB)

	Country	Number	Kind	Date	
Patent	EP	1194924	A1	20020410	(Basic)
	WO	200011660		20000302	
Application	EP	99948061		19990824	
	WO	99US19568		19990824	
Priorities	US	97569	P	19980824	
	US	156826		19980918	

Designated States:
DE; FR; GB;

International Patent Class (V7): G10L-019/14; G10L-019/14
NOTE: No A-document published by EPO

Type	Pub. Date	Kind	Text

Publication: English
Procedural: English
Application: English

Available Text	Language	Update	Word Count
CLAIMS B	(English)	200652	439
CLAIMS B	(German)	200652	346
CLAIMS B	(French)	200652	482
SPEC B	(English)	200652	19704
Total Word Count (Document A) 0			
Total Word Count (Document B) 20971			
Total Word Count (All Documents) 20971			

Specification: ...coding, waveform matching in the high frequency region proves more difficult than matching in the **low frequency** region. Thus, the **energy** of the high frequency region of a synthesized signal drops more than in the low...the PP mode, the input original signal has been pitch-preprocessed to match the interpolated **pitch contour**, so no closed-loop search is needed. The LTP excitation vector is computed using the interpolated **pitch contour** and the past synthesized excitation.

Fourth, the encoder processing circuitry generates a new target signal... ...6.65 kbps, the decision algorithm is as follows. First, at the block 241, a **prediction** of the **pitch** lag pit for the current frame is determined as follows: if(LTP(underscore)MODE(underscore... ...I m - 1.0 . The obtained index Im)) will be sent to the decoder.

The **pitch** lag **contour**, tau c)) (n), is defined using both the current lag Pm)) and the previous lag... ...the past modified weighted speech buffer. s(circumflex) w))(m0+n), n<0, with the **pitch** lag **contour**, tauc)) (n+m.Ls))), m=0.1.2. where Tc))(n) and TIC))(n) are... ...search, and LTP excitation is directly computed according to past synthesized excitation because the interpolated **pitch contour** is set for each frame. When the AMR coder operates with LTP-mode, the pitch... ...n<L(underscore)SF), is calculated by interpolating the past excitation (adaptive codebook) with the **pitch** lag **contour** , t r)) (n + m.L(underscore)SF), m = 0,1,2,3. The interpolation is... the innovation codebook gain) are coded for every subframe. The LSF vector is coded using **predictive** vector quantization. The **pitch** lag has an integer part and a fractional part constituting the pitch period. The quantized... ...result, for example, a codec might produce a synthesized residual that has greater high frequency **energy** and lesser **low frequency energy** than would otherwise be desired. In other words, the resultant synthesized residual would exhibit an...

11/3K/4 (Item 1 from file: 349) Links
PCT FULLTEXT

01304476
SYSTEM AND METHOD FOR ANALYZING AND IMPROVING A DISCOURSE ENGAGED IN BY A NUMBER OF INTERACTING AGENTS
SYSTEME ET PROCEDE PERMETTANT D'ANALYSER ET AMELIORER UN DISCOURS ENTRE UN CERTAIN NOMBRE D'AGENTS INTERACTIFS

Patent Applicant/Patent Assignee:

- **QUALIA INC**; 3 Bow Street, 4th Floor, Cambridge, MA 02138-5103
  US; US(Residence); US(Nationality)
  (For all designated states except: US)
- **PETTINELLI Eugene**; 110 Prides Crossing, Sudbury, MA 01776
  US; US(Residence); US(Nationality)
  (Designated only for: US)
- **ALEXANDER Daniel**; 38 Vermont Street, West Roxbury, MA 02132
  US; US(Residence); US(Nationality)
  (Designated only for: US)

Patent Applicant/Inventor:

- **PETTINELLI Eugene**
  110 Prides Crossing, Sudbury, MA 01776; US; US(Residence); US(Nationality); (Designated only for: US)
- **ALEXANDER Daniel**
  38 Vermont Street, West Roxbury, MA 02132; US; US(Residence); US(Nationality); (Designated only for: US)

Legal Representative:

- **KELLY Edward J(et al)(agent)**
  Ropes & Gray LLP, One International Place, Boston, MA 02110-2624; US;

	Country	Number	Kind	Date
Patent	WO	2005111999	A2	20051124
Application	WO	2005US13605		20050421

Priorities	US	2004566482		20040429
	US	20045872		20041206

Designated States: (All protection types applied unless otherwise stated - for applications 2004+)
AE; AG; AL; AM; AT; AU; AZ; BA; BB; BG;
BR; BW; BY; BZ; CA; CH; CN; CO; CR; CU;
CZ; DE; DK; DM; DZ; EC; EE; EG; ES; FI;
GB; GD; GE; GH; GM; HR; HU; ID; IL; IN;
IS; JP; KE; KG; KM; KP; KR; KZ; LC; LK;
LR; LS; LT; LU; LV; MA; MD; MG; MK; MN;
MW; MX; MZ; NA; NI; NO; NZ; OM; PG; PH;
PL; PT; RO; RU; SC; SD; SE; SG; SK; SL;
SM; SY; TJ; TM; TN; TR; TT; TZ; UA; UG;
US; UZ; VC; VN; YU; ZA; ZM; ZW;

[EP] AT; BE; BG; CH; CY; CZ; DE; DK; EE; ES;
FI; FR; GB; GR; HU; IE; IS; IT; LT; LU;
MC; NL; PL; PT; RO; SE; SI; SK; TR;

[OA] BF; BJ; CF; CG; CI; CM; GA; GN; GQ; GW;
ML; MR; NE; SN; TD; TG;

[AP] BW; GH; GM; KE; LS; MW; MZ; NA; SD; SL;
SZ; TZ; UG; ZM; ZW;

[EA] AM; AZ; BY; KG; KZ; MD; RU; TJ; TM;

Main International Patent Classes (Version 7):

IPC	Level
G10L-011/04	Main

Publication Language: English
Filing Language: English
Fulltext word count: 11954

Detailed Description:

...Parameters Measured from a
Speech Signal: PITCH-RELATED PROSODIC CUES
Pitch or fundamental frequency FO, **Pitch contour** (possibly smoothened), Mean FO, Median FO, Maximum FO, Minimum FO, FO range, about 95 th... ...to verbal prosody, that is, the set of suprasegmental features of speech, such as stress, **pitch**, **contour**, juncture, intonation (melody), rhythm, tempo, loudness, voice quality (smooth, coarse, shaky, creaky phonation, grumbly, etc... ...to separate speech
waveforms associated with various speakers (and optionally from ambient
sounds) using a **low-frequency energy**-based scheme (T. Choudhury and A.

Pentland, "Modeling Face-to-Face Communication Using the Sociometer...

11/3K/5 (Item 2 from file: 349) Links
PCT FULLTEXT

01110815
METHODS FOR INTEROPERATION BETWEEN ADAPTIVE MULTI-RATE WIDEBAND (AMR-WB) AND MULTI-MODE VARIABLE BIT-RATE WIDEBAND (WMR-WB) SPEECH CODECS
PROCEDE D'INTERFONCTIONNEMENT ENTRE CODEURS-DECODEURS LARGE BANDE DEBITS MULTIPLES ADAPTATIFS (AMR-WB) ET CODEURS-DECODEURS LARGE BANDE DEBIT BINAIRE VARIABLE MULTIMODES (VMR-WB)

Patent Applicant/Patent Assignee:

- **VOICEAGE CORPORATION**; 750 chemin Lucerne, Suite 250, Ville Mont-Royal, Quebec H3R 2H6
  CA; CA(Residence); CA(Nationality)
  (For all designated states except: US)
- **JELINEK Milan**; 925 Walton, Sherbrooke, Quebec J1H 1K4
  CA; CA(Residence); CA(Nationality)
  (Designated only for: US)
- **SALAMI Redwan**; 4045 Place Albert-Dreux, Ville St-Laurent, Quebec H4R 2Y3
  CA; CA(Residence); CA(Nationality)
  (Designated only for: US)

Patent Applicant/Inventor:

- **JELINEK Milan**
  925 Walton, Sherbrooke, Quebec J1H 1K4; CA; CA(Residence); CA(Nationality); (Designated only for: US)
- **SALAMI Redwan**
  4045 Place Albert-Dreux, Ville St-Laurent, Quebec H4R 2Y3; CA; CA(Residence); CA(Nationality); (Designated only for: US)

Legal Representative:

- **BROUILLETTE Robert(et al)(agent)**
  Brouillette Kosie Prince, 1100 Rene-Levesque Blvd. West, 25th Floor, Montreal, Quebec H3B 5C9; CA;

	Country	Number	Kind	Date
Patent	WO	200434376	A2-A3	20040422
Application	WO	2003CA1572		20031010
Priorities	US	2002417667		20021011

Designated States: (All protection types applied unless otherwise stated - for applications 2004+)

[EP] AT; BE; BG; CH; CY; CZ; DE; DK; EE; ES;
FI; FR; GB; GR; HU; IE; IT; LU; MC; NL;
PT; RO; SE; SI; SK; TR;

[OA] BF; BJ; CF; CG; CI; CM; GA; GN; GQ; GW;
ML; MR; NE; SN; TD; TG;

[AP] GH; GM; KE; LS; MW; MZ; SD; SL; SZ; TZ;
UG; ZM; ZW;

[EA] AM; AZ; BY; KG; KZ; MD; RU; TJ; TM;

Main International Patent Classes (Version 7):

IPC	Level
G10L-019/14	Main

Publication Language: English
Filing Language: English
Fulltext word count: 16813

Detailed Description:

...200, the spectral tilt is estimated in the frequency domain as a ratio between the **energy** concentrated in **low frequencies** and the **energy** concentrated in high frequencies. However, it can be also estimated in different ways such as... ...and j, is the index of the first bin in the ith critical band.

The **energy** in **low frequencies** is computed as the average of the energies.in the first 1 0 critical bands... ...bands have been excluded from the computation to improve the discrimination between frames with high-**energy** concentration in **low frequencies**

(generally voiced) and with high-energy concentration in high frequencies (generally unvoiced). In between, the ...
...content is not characteristic for any of the classes and increases the decision confusion.

The **energy** in **low frequencies** is computed differently for
long pitch periods and short pitch periods. For voiced female speech... ...to the nearest harmonics are taken into account. Hence, if the structure is harmonic in **low frequencies**, only high-**energy** terms will 1 5 be included in the sum. On the other hand, if the... ...will be random and the sum will be smaller.

Thus even unvoiced sounds with high **energy** content in **low frequencies**
can be detected. This processing cannot be done for longer pitch periods, as the frequency... ...sufficient. For pitch values larger than .20 128 or for a priori unvoiced sounds the **low frequency energy** is computed
per critical band as
9
El EcB(k)
10 k=O
A priori... Interpolation gives a delay value for every time instant of the frame.

After the delay **contour** is available, the **pitch** in the subframe to be coded currently is adjusted to follow this artificial contour by... ...other signal-coding parameters. In signal modification, the signal is forced to follow a certain **pitch contour** that can be transmitted
with 9 bits per frame. Good performance of long-term prediction...

11/3K/6 (Item 3 from file: 349) Links
PCT FULLTEXT

01110814
METHODS AND DEVICES FOR SOURCE CONTROLLED VARIABLE BIT-RATE WIDEBAND SPEECH CODING
PROCEDES ET DISPOSITIFS DE CODAGE VOCAL LARGE BANDE EN DEBIT BINAIRE VARIABLE COMMANDE PAR LA SOURCE

Patent Applicant/Patent Assignee:

- **VOICEAGE CORPORATION**; 750 chemin Lucerne, Suite 250, Ville Mont-Royal, Quebec H3R 2H6
  CA; CA(Residence); CA(Nationality)
  (For all designated states except: US)
- **JELINEK Milan**; 925 Walton, Sherbrooke, Quebec J1H 1K4
  CA; CA(Residence); CA(Nationality)
  (Designated only for: US)

Patent Applicant/Inventor:

- **JELINEK Milan**
  925 Walton, Sherbrooke, Quebec J1H 1K4; CA; CA(Residence); CA(Nationality); (Designated only for: US)

Legal Representative:

- **BROUILLETTE Robert(et al)(agent)**
  Brouillette Kosie Prince, 1100 Rene-Levesque Blvd. West, 25th Floor, Montreal, Quebec, H3B 5C9; CA;

	Country	Number	Kind	Date
Patent	WO	200434379	A2-A3	20040422
Application	WO	2003CA1571		20031009
Priorities	US	2002417667		20021011

Designated States: (All protection types applied unless otherwise stated - for applications 2004+)

|EP| AT; BE; BG; CH; CY; CZ; DE; DK; EE; ES;
FI; FR; GB; GR; HU; IE; IT; LU; MC; NL;
PT; RO; SE; SI; SK; TR;

|OA| BF; BJ; CF; CG; CI; CM; GA; GN; GQ; GW;
ML; MR; NE; SN; TD; TG;

|AP| GH; GM; KE; LS; MW; MZ; SD; SL; SZ; TZ;
UG; ZM; ZW;

|EA| AM; AZ; BY; KG; KZ; MD; RU; TJ; TM;

Main International Patent Classes (Version 7):

IPC	Level
G10L-019/14	Main

Publication Language: English
Filing Language: English
Fulltext word count: 17155

Detailed Description:

...200, the spectral tilt is
estimated in the frequency domain as a ratio between the **energy**
concentrated in **low frequencies** and the **energy** concentrated in high
frequencies. However, it can be also estimated in different ways such as... ...and j, is the index of the first bin in the ith critical band.

The **energy** in **low frequencies** is computed as the average of
the energies in the first 10 critical bands. The... ...been excluded from the computation to improve the discrimination
I 0 between frames with high-**energy** concentration in **low frequencies**
(generally voiced) and with high-energy concentration in high frequencies (generally unvoiced). In between, the ...
...characteristic for any of the classes and increases the decision confusion.

1 5 i The **energy** in **low frequencies** is computed differently for
long pitch periods and short pitch.periods. For voiced female speech... ...to the nearest harmonics are taken into account. Hence, if the structure is harmonic in **low frequencies**, only high-**energy** terms will be included in the sum. On the other hand, if the structure is... ...will be random and the sum will be smaller.

Thus even unvoiced sounds with high **energy** content in **low frequencies**
can be detected. This processing cannot be done for longer pitch periods, I 0 as... ...not sufficient. For pitch values larger than
128 or for a priori unvoiced sounds the **low frequency energy** is computed
per critical band as
1
El =-YEcB(k)
j
10 k=O
1...Interpolation gives a delay value for every time instant of the frame.

After the delay **contour** is available, the **pitch** in the subframe to be coded currently is adjusted to follow this artificial contour by... ...other signal-coding parameters. In signal modification, the signal is forced to follow a certain **pitch contour** that can be transmitted
with 9 bits per frame. Good performance of long-term prediction...

Claims:

...recited in claim 7, wherein said spectral tilt
is proportionate to a ratio between the **energy** concentrated in **low frequencies** and the **energy** concentrated in high frequencies of said signal frame.
9 A method as recited in claim 8, wherein said **energy**
concentrated in **low frequencies** and said **energy** concentrated in highfrequencies are computed following the perceptual critical bands.1 0. A method... ...recited in claim 24, wherein said spectral tiltis proportionate to a ratio

between the **energy** concentrated in **low frequencies** and the **energy** concentrated in high frequencies of said signal frame.26.A method as recited in claim 25, wherein said **energy**concentrated in **low frequencies** and said **energy** concentrated in highfrequencies are computed following the perceptual critical bands.
27 A method as...

11/3K/7 (Item 4 from file: 349) **Links**
PCT FULLTEXT

00784165
A SYSTEM, METHOD, AND ARTICLE OF MANUFACTURE FOR A TELEPHONIC EMOTION DETECTOR THAT PROVIDES OPERATOR FEEDBACK
SYSTEME, PROCEDE, ET ARTICLE DE FABRICATION DESTINE A UN DETECTEUR D'EMOTIONS TELEPHONIQUE FOURNISSANT UN RETOUR A L'OPERATEUR

Patent Applicant/Patent Assignee:

- **ANDERSEN CONSULTING LLP**; 1661 Page Mill Road, Palo Alto, CA 94304
  US; US(Residence); US(Nationality)

Legal Representative:

- **HICKMAN Paul L(agent)**
  Hickman Coleman & Hughes, LLP, P.O. Box 52037, Palo Alto, CA 94303-0746; US;

	Country	Number	Kind	Date
Patent	WO	200116939	A1	20010308
Application	WO	2000US24325		20000831
Priorities	US	99387621		19990831

Designated States: (All protection types applied unless otherwise stated - for applications 2004+)

[EP] AT; BE; CH; CY; DE; DK; ES; FI; FR; GB;
GR; IE; IT; LU; MC; NL; PT; SE;

[OA] BF; BJ; CF; CG; CI; CM; GA; GN; GW; ML;
MR; NE; SN; TD; TG;

[AP] GH; GM; KE; LS; MW; MZ; SD; SL; SZ; TZ;
UG; ZW;

[EA] AM; AZ; BY; KG; KZ; MD; RU; TJ; TM;

Main International Patent Classes (Version 7):

IPC	Level
G10L-017/00	Main

Publication Language: English
Filing Language: English
Fulltext word count: 38681

Detailed Description:

...features such as LPC (linear predictive coding) parameters of signal or features of the smoothed **pitch contour** and its derivatives.

For this invention, the following strategy may be adopted. First, take into... ...a linear regression for voiced part of speech, i.e. the line that fits the **pitch contour**. The relative voiced energy can also be calculated as the proportion of

voiced energy to...in effect correspond to resonances of the vocal tract, the human voice also contains a **pitch**, **modulated** by the speaker, which corresponds to the frequency at which the larynx modulates the air...speech to text systems.

The first reflection coefficient ki is approximately related to the high/**low frequency energy** ratio and a signal. See R. J. McAulay, "Design of a Robust Maximum Likelihood Pitch ... ...1979, which is hereby incorporated by reference. For ki close to -1, there is more **low frequency energy** in the signal than high-frequency energy, and vice versa for k, close to 1...their goodness values C(k), dynamic programming is now
57
used to obtain an optimum **pitch contour** which includes an optimum voicing decision for each ftame. The dynamic programming requires several frames...

11/3K/8 (Item 5 from file: 349) Links
PCT FULLTEXT

00784164
A SYSTEM, METHOD, AND ARTICLE OF MANUFACTURE FOR DETECTING EMOTION IN VOICE SIGNALS THROUGH ANALYSIS OF A PLURALITY OF VOICE SIGNAL PARAMETERS
SYSTEME, PROCEDE ET ARTICLE MANUFACTURE DE DETECTION DES EMOTIONS DANS LES SIGNAUX VOCAUX PAR ANALYSE D'UNE PLURALITE DE PARAMETRES DE SIGNAUX VOCAUX

Patent Applicant/Patent Assignee:

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- **HICKMAN Paul L(agent)**
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	Country	Number	Kind	Date
Patent	WO	200116938	A1	20010308
Application	WO	2000US23884		20000831
Priorities	US	99388027		19990831

Designated States: (All protection types applied unless otherwise stated - for applications 2004+)

[EP] AT; BE; CH; CY; DE; DK; ES; FI; FR; GB;
GR; IE; IT; LU; MC; NL; PT; SE;

[OA] BF; BJ; CF; CG; CI; CM; GA; GN; GW; ML;
MR; NE; SN; TD; TG;

[AP] GH; GM; KE; LS; MW; MZ; SD; SL; SZ; TZ;
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[EA] AM; AZ; BY; KG; KZ; MD; RU; TJ; TM;

Main International Patent Classes (Version 7):

IPC	Level
G10L-017/00	Main

Publication Language: English
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Fulltext word count: 39489

Detailed Description:

...features such as LPC (linear predictive coding) parameters of signal or features of the smoothed **pitch contour** and its derivatives.

For this invention, the following strategy may be adopted. First, take into... ...a linear regression for voiced part of speech, i.e. the line that fits the **pitch contour**. The relative voiced energy can also be calculated as the proportion of voiced energy to...in effect correspond to resonances of the vocal tract, the human voice also contains a **pitch, modulated** by the speaker, which corresponds to the frequency at which the larynx modulates the air...speech to text systems.

The first reflection coefficient ki is approximately related to the high/**low frequency energy** ratio and a signal. See R. J. McAulay, "Design of a Robust Maximum Likelihood Pitch ... ...1979, which is hereby incorporated by reference. For k, close to - 1, there is more **low frequency energy** in the signal than high-frequency energy, and vice versa for k, close to 1...their goodness values C(k), dynamic programming is now
57
used to obtain an optimum **pitch contour** which includes an optimum voicing decision for each frame. The dynamic programming requires several frames...

11/3K/9 (Item 6 from file: 349) Links
PCT FULLTEXT

00783383
VOICE RECOGNITION FOR INTERNET NAVIGATION
RECONNAISSANCE VOCALE POUR NAVIGATION INTERNET

Patent Applicant/Patent Assignee:

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Patent	WO	200116936	A1	20010308
Application	WO	2000US24302		20000831
Priorities	US	99387195		19990831

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[OA] BF; BJ; CF; CG; CI; CM; GA; GN; GW; ML;
MR; NE; SN; TD; TG;

[AP] GH; GM; KE; LS; MW; MZ; SD; SL; SZ; TZ;
UG; ZW;

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Main International Patent Classes (Version 7):

IPC	Level

G10L-015/26	Main

Publication Language: English
Filing Language: English
Fulltext word count: 38996

Detailed Description:

...as LPC (linear predictive coding) parameters of signal or features of the 1 5 smoothed **pitch contour** and its derivatives.

For this invention, the following strategy may be adopted. First, take into ... ...a linear regression for voiced part of speech, i.e. the line that fits the **pitch contour**. The relative voiced energy can also be calculated as the proportion of voiced energy to...in effect correspond to resonances of the vocal tract, the human voice also contains a **pitch, modulated** by the speaker, which corresponds to the frequency at which the larynx modulates the air...speech to text systems.

The first reflection coefficient k, is approximately related to the high/**low frequency energy** ratio and a signal. See R. J. McAulay, "Design of a Robust Maximum Likelihood Pitch ... ...1979, which is hereby incorporated by reference. For k, close to -1, there is more **low frequency energy** in the signal than high-frequency energy, and vice versa for k, close to 1...their goodness values C(k), dynamic programming is. now
58
used to obtain an optimum **pitch contour** which includes an optimum voicing decision for each frame. The dynamic programming requires several frames...

11/3K/10 (Item 7 from file: 349) Links
PCT FULLTEXT

00783339
SYSTEM, METHOD, AND ARTICLE OF MANUFACTURE FOR A BORDER CROSSING SYSTEM THAT ALLOWS SELECTIVE PASSAGE BASED ON VOICE ANALYSIS
SYSTEME, PROCEDE ET ARTICLE MANUFACTURE POUR SYSTEME DE FRANCHISSEMENT DE FRONTIERE PERMETTANT UN PASSAGE SELECTIF SUR LA BASE DE L'ANALYSE DE LA VOIX

Patent Applicant/Patent Assignee:

- **ANDERSEN CONSULTING LLP**; 1661 Page Mill Road, Palo Alto, CA 94304
  US; US(Residence); US(Nationality)

Legal Representative:

- **HICKMAN Paul L(agent)**
  Hickman Coleman & Hughes, LLP, P.O. Box 52037, Palo Alto, CA 94303-0746; US;

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Patent	WO	200116892	A1	20010308
Application	WO	2000US24313		20000831
Priorities	US	99387415		19990831

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[EP] AT; BE; CH; CY; DE; DK; ES; FI; FR; GB;
GR; IE; IT; LU; MC; NL; PT; SE;

[OA] BF; BJ; CF; CG; CI; CM; GA; GN; GW; ML;
MR; NE; SN; TD; TG;

[AP] GH; GM; KE; LS; MW; MZ; SD; SL; SZ; TZ; UG; ZW;

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Main International Patent Classes (Version 7):

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Publication Language: English
Filing Language: English
Fulltext word count: 39788

Detailed Description:

...features such as LPC (linear predictive coding) parameters of signal or features of the smoothed **pitch contour** and its derivatives.

For this invention, the following strategy may be adopted. First, take into...a linear regression for voiced part of speech, i.e. the line that fits the **pitch contour**. The relative voiced energy can also be calculated as the proportion of voiced energy to... in effect correspond to resonances of the vocal tract, the human voice also contains a **pitch, modulated** by the speaker, which corresponds to the frequency at which the larynx modulates the air... speech to text systems.

The first reflection coefficient ki is approximately related to the high/**low frequency energy** ratio and a signal. See R. J. McAulay, "Design of a Robust Maximum Likelihood Pitch... ...1979, which is hereby incorporated by reference. For k, close to -1, there is more **low frequency energy** in the signal than high-frequency energy, and vice versa for k, close to 1...their goodness values C(k), dynamic programming is now
59
used to obtain an optimum **pitch contour** which includes an optimum voicing decision for each frame. The dynamic programming requires several frames...

11/3K/11 (Item 8 from file: 349) Links
PCT FULLTEXT

00548287
ADAPTIVE TILT COMPENSATION FOR SYNTHESIZED SPEECH RESIDUAL
COMPENSATION D'INCLINAISONS ADAPTATIVE POUR RESIDUS VOCAUX SYNTHETISES

Patent Applicant/Patent Assignee:

- **CONEXANT SYSTEMS INC;**
 ; ;

	Country	Number	Kind	Date
Patent	WO	200011660	A1	20000302
Application	WO	99US19568		19990824
Priorities	US	9897569		19980824
	US	98156826		19980918

Designated States: (All protection types applied unless otherwise stated - for applications 2004+)

Main International Patent Classes (Version 7):

IPC	Level
G10L-019/14	Main

Publication Language: English
Filing Language:
Fulltext word count: 28380

Detailed Description:

...coding, waveform matching in the high frequency region proves more difficult than matching in the **low frequency** region. Thus, the **energy** of the high frequency region of a synthesized speech signal drops more than in the...the PP mode, the input original signal has been pitch-preprocessed to match the interpolated **pitch contour**, so no closed-loop search is needed. The UP excitation vector is computed using the interpolated **pitch contour** and the past synthesized excitation.

Fourth, the encoder processing circuitry generates a new target signal...65 kbps, the decision algorithm is as follows. First, at the block 24 1, a **prediction** of the **pitch** lag pit for the current frame is determined as follows.

if (UP-MODE-M == 1...max (I,,, - 1, 0).

The obtained index I,, will be sent to the decoder.

The **pitch** lag **contour**, rc(n), is defined using both the current lag P" and the previous lag Pn... ...by warping the past modified weighted speech buffer, 9,,(mO + n), n < 0, with the **pitch** lag **contour**, -r, (n + m - Ls), m = 0,1,2,
fj
S,, (mO + n) S,,, (mO + n...search, and UP excitation is directly computed according to past synthesized excitation because the interpolated **pitch contour** is set for each frame. When the AMR coder operates with LTP-mode, the pitch... ...0<=n<L,
SFI, is calculated by interpolating the past excitation (adaptive
codebook) with the **pitch** lag **contour**, r. (n + m-L
SF), m = 0.1,2,3. The interpolation is
performed using...the innovation codebook gain) are coded for every subframe. The LSF vector is coded using **predictive** vector quantization. The **pitch** lag has an integer part and a fractional part constituting the pitch period. The quantized...result, for example, a codec might produce a synthesized residual that has greater high frequency **energy** and lesser **low frequency energy** than would otherwise be desired. In other words, the resultant synthesized residual would exhibit an...

13/3K/1 (Item 1 from file: 348) Links
EUROPEAN PATENTS

01563968
A method of bandwidth extension for narrow-band speech
Verfahren zur Erweiterung der Bandbreite eines schmalbandigen Sprachsignals
Procede pour l'extension de la largeur de bande d'un signal vocal a bande etroite

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  (Proprietor designated states: all)

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Legal Representative:

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  Marks & Clerk, 4220 Nash Court, Oxford Business Park South; Oxford,Oxfordshire OX4 2RU; (GB)

	Country	Number	Kind	Date	
Patent	EP	1300833	A2	20030409	(Basic)
	EP	1300833	A3	20050216	
	EP	1300833	B1	20061122	
Application	EP	2002257102		20021004	
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Designated States:
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AL; LT; LV; MK; RO; SI;

International Patent Class (V7): G10L-021/02; G10L-021/02

IPC	Level	Value	Position	Status	Version	Action	Source	Office
G10L-0021/02	A	I	F	B	20060101	20021211	H	EP
G10L-0021/02	A	I	F	B	20060101	20021211	H	EP

Abstract ...comprises synthesizing a wideband signal using the wideband LPCs and a wideband excitation signal, highpass **filtering** the synthesized wideband signal to produce a highband signal, and combining the highband signal with...

Abstract Word Count: 187
NOTE: 8
NOTE: Figure number on first page: 8

Type	Pub. Date	Kind	Text

Publication: English
Procedural: English
Application: English

Available Text	Language	Update	Word Count
CLAIMS A	(English)	200315	1758
SPEC A	(English)	200315	13164
CLAIMS B	(English)	200647	1865
CLAIMS B	(German)	200647	1704
CLAIMS B	(French)	200647	2224
SPEC B	(English)	200647	13195
Total Word Count (Document A) 14926			
Total Word Count (Document B) 18988			
Total Word Count (All Documents) 33914			

Specification: ...and determining its amplitude based on the amplitudes of the surrounding narrowband samples via lowpass **filtering**. However, there is weakness in the interpolated speech in that it does not contain any... ...by a factor of 2 by inserting a zero sample following each input sample, highpass **filtering** with additional spectral shaping 38, and gain adjustment 40. Since the spectral folding operation reflects... ...lower band into the upper band, i.e., highband, the purpose of the spectral shaping **filter** is to attenuate these signals in the highband. To reduce the spectral-gap about 4kHz... ...in the art. See, e.g., H. Yasukawa, Quality Enhancement of Band Limited Speech by **Filtering** and Multirate Techniques, in Proc. Intl. Conf. Spoken Language Processing, ICSLP '94, pp. 1607-1610... ...input signal. Preferably, fullwave rectification is used for this purpose. Again, highpass and spectral shaping **filters** 48 with a gain adjustment 50 are applied to the rectified signal to generate the ... ...a wideband excitation signal, to be shaped by the generated wideband spectral envelope 58. Highpass **filtering** and gain 60 extract the highband signal for combining with the original narrowband signal to......logarithmic, typically extracted from an LP model. Almost all parametric techniques use an LPC synthesis **filter** for wideband signal generation (typically an intermediate wideband signal which is further highpass **filtered**), by exciting it with an appropriate wideband excitation signal.

Parametric methods can be further classified... ...for synthesizing the highband signal. The synthesis is carried out by exciting the LPC synthesis **filter** by a wideband excitation signal. The excitation signal is obtained by inverse **filtering** the input narrowband signal and spectral folding the resulting residual signal. The main disadvantage of... ...air turbulences at constrictions in the vocal tract provide the excitation for unvoiced sounds. By **filtering** the speech signal with an inverse **filter**, whose coefficients are determined form the LPC model, the effect of the formants is removed... ...LPC coefficients are used for synthesizing a wideband signal. The synthesized wideband signal is highpass **filtered** and summed with the original narrowband signal to generate the output wideband signal. Any monotonic... ...The narrowband module comprises a signal interpolation module producing an interpolated narrowband signal, an inverse **filter** that **filters** the interpolated narrowband signal and a nonlinear operation module that generates an excitation signal from the **filtered** interpolated narrowband signal. The system further comprises a module for producing wideband coefficients. The wideband... ...the wideband coefficients and the wideband excitation signal to synthesize a wideband signal. A highpass **filter** and gain module **filters** the wideband signal and adjusts the gain of the resulting highband signal. A summer sums... ...using the wideband LPCs, and a wideband excitation signal generated from the narrowband signal; highpass **filter** the synthesized wideband signal to generate the synthesized highband signal; and sum the synthesized highband... ...LPCs, synthesizing a wideband signal using the wideband LPCs and a wideband residual signal, highpass **filtering** the synthesized wideband signal to generate a synthesized highband signal, and generating the wideband signal... ...of the present invention;

Fig. 9 shows the frequency response of a low pass interpolation **filter**;

Fig. 10 shows the frequency response of an Intermediate Reference System (IRS), an IRS compensation **filter** and the cascade of the two;

Fig. 11 is a flowchart representing an exemplary method... ...parameters. In the training phase, wideband speech signals and the corresponding narrowband signals, obtained bylowpass **filtering**, are available so that the relationship between the corresponding parameter sets could be determined.

Some... ...synthesizing a wideband signal using an LPC synthesis approach followed by highpass and spectral shaping **filters**. The method according to the present invention also belongs to this category of parametric with... ...converted first to LP parameters. These LP parameters are then used to construct a synthesis **filter**, which needs to be excited by a suitable wideband excitation signal.

Two alternative approaches, commonly... ...and 5B. First, as shown in Fig. 5A, the narrowband input speech signal is inverse **filtered** 72 using previously extracted LP coefficients to obtain a narrowband residual signal. This is accomplished... ...flattening can be done by applying an LPC analysis to this signal, followed by inverse **filtering**.

A second and preferred alternative is shown in Fig. 5B. It is useful for reducing... ...need to perform the necessary additional interpolation in the first scheme. To perform the inverse **filtering** 84, the option exists in this case for either using the wideband LP parameters obtained from the mapping stage to get the inverse **filter** coefficients, or inserting zeros, like in spectral folding, into the narrowband LP coefficient vector. The... ...when a nonlinear operator is used, i.e., using the original LP coefficients for inverse **filtering** 72 the input narrowband signal followed by interpolation. The bandwidth of the resulting residual signal...637-655, 1971; and H. Wakita, Direct Estimation of the Vocal Tract Shape by Inverse **Filtering** of Acoustic Speech Waveform, IEEE Trans. Audio and Electroacoust., vol. AU-21, No. 5, pp ... ...sections of equal length, as schematically shown in Fig. 6. Moreover, an equivalence of the **filtering** process by the acoustic tube and by the LP all-pole **filter** model of the pre-emphasized speech has been shown to exist under the constraint: In... ...to compensate for the glottal wave shape and lip radiation. Typically, a fixed pre-emphasis **filter** is used, usually of the form 1 - (mu)z-1), where (mu) is chosen to... ...of the original wideband speech signal from which the narrowband signal was extracted by lowpass **filtering**. Using the approach according to the present invention, one can find a refinement as demonstrated... ...byupsampling 112, for example, by inserting a zero sample following each input sample and lowpass **filtering** at 4 kHz, yielding the narrowband interpolated signal . The symbol "(equivalent to)" relates to narrowband interpolated signals. Because of the spectral folding caused by upsampling, high **energy** formants at **low frequencies**, typically present in voiced speech, are reflected to high frequencies and need to be strongly attenuated by the lowpass **filter** (not shown). Otherwise, relatively strong undesired signals may appear in the synthesized highband.

Preferably, the lowpass **filter** is designed using the simple window method for FIR **filter** design, using a window function with sufficiently high sidelobes attenuation, like the Blackman window. See... ...increases with frequency, as desired here. The frequency response of a 129 long FIR lowpass **filter** designed with a Blackman window and used in simulations is shown in Fig. 9.

In... ...frame update is used. The signal is first pre-emphasized using a first order FIR **filter** 1 - (mu)z-1), with (mu) = (rho)1)), where, as mentioned above, (rho)1)) is... ...in this paragraph are all performed by the LPC analysis module 114. The corresponding inverse **filter** transfer function is then given by Anb)) (z) : However, to generate the LPC residual signal... ...higher sampling rate (fwbs = 16 kHz if fnbs = 8 kHz), the interpolated signal is inverse **filtered** by Anb)) (z2)), as shown by block 126. The **filter** coefficients, which are denoted by anb) (up arrow) 2, are simply obtained from anb) by... ...i.e., inserting zeros - as done for spectral folding. Thus, the coefficients of the inverse **filter** Anb)) (z2)), operating at the high sampling frequency, including the unity leading term, are: The... ...Mnb)), not to be confused with Anb)) (z) in equ. (3), which denotes the inverse-**filter** transfer function, are computed 116 from the partial correlation coefficients (parcors) of the narrowband signal... ...coefficients represent a wideband spectral envelope.

To synthesize the highband signal, the wideband LPC synthesis **filter** 122, which uses these coefficients, needs to be excited by a signal that has energy... ...the whole upper band, is a desired feature and eliminates the need to apply any **filtering** in addition to highpass **filtering** 134. Fullwave rectification is preferred. A memoryless nonlinearity maintains signal periodicity, thus avoiding artifacts caused... ...present invention also takes into account that the highband signal of natural wideband speech has **pitch** dependent time-envelope **modulation**, which is preserved by the nonlinearity. The inventor's preference of fullwave rectification over the... ...of spectral tilt is desired, then either the wideband excitation can be flattened via inverse **filtering**, as discussed above, or infinite clipping can be used having the characteristics shown in Fig... ...is not identical to the original input narrowband signal, the synthesized signal is preferably highpass **filtered** 134 and the resulting highband signal, Shb)), is gain adjusted 134 and added 136 to ... ...put signal S(sup AND)wb)). Note that like the gain factor, also the highpass **filter** can be applied either before or after the wideband LPC synthesis block.

While Fig. 8... ...done in Fig. 8 (the HPF needs then to be replaced by a proper shaping **filter** to attenuate high frequencies, as discussed earlier). The use of spectral folding is, of course... ...Fig. 8 on the above residual signal (i.e., obtained by using awb)), but highpass

filter its output, and combine it (after proper gain adjustment) with the interpolated narrowband residual signal... ...the wideband excitation signal rwb)). This signal is fed then into the wideband LPC synthesis **filter**. Here again the resulting signal, ywb)), can serve as the desired output signal.

Various components... ...narrowband module from Fig. 8 may comprise the 1:2 interpolation block 112, the inverse **filter** 126 and the elements 128, 130 and 132 to generate an excitation signal from the... ...signal.

Another way to generate a highband signal is to excite the wideband LPC synthesis **filter** (constructed from the wideband LPC coefficients) by white noise and apply highpass **filtering** to the synthesized signal. While this is a well-known simple technique, it suffers from... ...Fig. 9 illustrates a graph 138 includes the frequency response of a low pass interpolation **filter** used for 2:1 signal interpolation. Preferably, the **filter** is a half-band linear-phase FIR **filter**, designed by the window method using a Blackman window.

When the narrowband speech is obtained... ...sector of the International Telecommunication Union), for analog telephone channels. The frequency response of a **filter** that simulates the IRS characteristics is shown in Fig. 10 as a dashed line 146... ...mitigate them. Also shown in Fig. 10 are the frequency response associated with a compensation **filter** 142 and the response associated with the cascade of the two (compensated response).

One aspect... ...to the above, and independently of it, it is useful to use an extended highpass **filter**, having a cutoff frequency Fc)) matched to the upper edge of the signal band (3... ...addition to an IRS channel response 146, Fig. 10 shows the response of a compensating **filter** 142 and the resulting compensated response 144, which is flat in the nominal range. The compensation **filter** designed here is an FIR **filter** of length 129. This number could be lowered even to 65, with only little effect. The compensated signal becomes then the input to the bandwidth extension system. This **filtering** of the output signal from a telephone channel would then be added as a block... ...art, the lowerband signal may be generated by just applying a narrow (300 Hz) lowpass **filter** to the synthesized wideband signal in parallel to the highpass **filter** 134 in Fig. 8. Other known work in the art addresses this issue more carefully... ...H. Yasukawa, Restoration of Wide Band Signal from Telephone Speech using Linear Prediction Residual Error **Filtering**, in Proc. IEEE Digital Signal Processing Workshop, pp. 176-178, 1996. This approach includes adding to the proposed system a 300 Hz LPF in parallel to the existing highpass **filter**. However, because the nonlinear operator injects also undesired components into the lowband (as excitation), audible...input signal, Snb)), by a factor, such as a factor of 2 (upsampling and lowpass **filtering**). This step results in a narrowband interpolated signal . The signal is inverse **filtered** (166) using, for example, a transfer function of Anb)) (z2)) having the coefficients shown in... ...rate.

Next, a non-linear operation is applied to the signal output from the inverse **filter**. The operation comprises fullwave rectification (absolute value) of residual signal (168). Other nonlinear operators discussed... ...to signal rectification (as discussed below) via LPC analysis of the rectified signal and inverse **filtering**. The preferred setting here is no spectral tilt compensation.

Next, the highband signal must be... ...added (174) to the original narrowband signal. This step comprises exciting a wideband LPC synthesis **filter** (170) (with coefficients awb)) by the generated wideband excitation signal rwb)), resulting in a wideband... ...may undergo further processing. If further processing is desired, the wideband signal ywb)) is highpass **filtered** (172) using a HPF having its cutoff frequency at Fc)) to generate a highband signal... ...original highband signal, which is maintained also in the generated highband signal.

Applying a dispersion **filter** such as an allpass nonlinear-phase **filter**, as in the 2400 bps DoD standard MELP coder, for example, can mitigate the spiky... ...wideband LPCs awbi and the wideband excitation signal, generating a highband signal Shb)) by highpass **filtering** ywb)), adjusting the gain and generating the wideband signal by summing the synthesized highband signal... ...a signal obtained bypassing a white Gaussian signal, v(n), through a half-band lowpass **filter** are discussed followed by some specific nonlinear memoryless operators, namely-generalized rectification, defined below, and... ...Hill, New York, 1965 ("Papoulis").

Referring to Fig. 18, the signal v(n) is lowpass **filtered** 320 to produce x(n) and then passed through a nonlinear operator 322 to produce a signal z(n). The lowpass **filtered** signal x(n) has, ideally, a flat spectral magnitude for -(pi) / 2 ≤ (theta) &le... ...v(n) has zero mean and variance (sigma)2v, and that the half-band lowpass **filter** is ideal, the autocorrelation functions of v(n) and x(n) are: where (delta)(m... ...in mitigating the 'spectral gap' near 4 kHz. It also helps when a narrow lowpass **filter** is used to extract from the synthesized wideband signal a synthetic lowband (0 - 300 Hz... ...to be useful. It can be added to the bandwidth extension system as a preprocessing **filter** at its input, as demonstrated herein.

It should be noted that when the input signal...

Specification: ...and determining its amplitude based on the amplitudes of the surrounding narrowband samples via lowpass **filtering**. However, there is weakness in the interpolated speech in that it does not contain any... ...by a factor of 2 by inserting a zero sample following each input sample, highpass **filtering** with additional spectral shaping 38, and gain adjustment 40. Since the spectral folding operation reflects... ...lower band into the upper band, i.e., highband, the purpose of the spectral shaping **filter** is to attenuate these signals in the highband. To reduce the spectral-gap about 4kHz... ...in the art. See, e.g., H. Yasukawa, Quality Enhancement of Band Limited Speech by **Filtering** and Multirate Techniques, in Proc. Intl. Conf. Spoken Language Processing, ICSLP '94, pp. 1607-1610... ...input signal. Preferably, fullwave rectification is used for this purpose. Again, highpass and spectral shaping **filters** 48 with a gain adjustment 50 are applied to the rectified signal to generate the ... ...a wideband excitation signal, to be shaped by the generated wideband spectral envelope 58. Highpass **filtering** and gain 60 extract the highband signal for combining with the original narrowband signal to......logarithmic, typically extracted from an LP model. Almost all parametric techniques use an LPC synthesis **filter** for wideband signal generation (typically an intermediate wideband signal which is further highpass **filtered**), by exciting it with an appropriate wideband excitation signal.

Parametric methods can be further classified... ...for synthesizing the highband signal. The synthesis is carried out by exciting the LPC synthesis **filter** by a wideband excitation signal. The excitation signal is obtained by inverse **filtering** the input narrowband signal and spectral folding the resulting residual signal. The main disadvantage of... ...air turbulences at constrictions in the vocal tract provide the excitation for unvoiced sounds. By **filtering** the speech signal with an inverse **filter**, whose coefficients are determined from the LPC model, the effect of the formants is removed...LPC coefficients are used for synthesizing a wideband signal. The synthesized wideband signal is highpass **filtered** and summed with the original narrowband signal to generate the output wideband signal. Any monotonic ... ...of the present invention;

Fig. 9 shows the frequency response of a low pass interpolation **filter**;

Fig. 10 shows the frequency response of an Intermediate Reference System (IRS), an IRS compensation **filter** and the cascade of the two;

Fig. 11 is a flowchart representing an exemplary method... ...In the training phase, wideband speech signals and the corresponding narrowband signals, obtained by lowpass **filtering**, are available so that the relationship between the corresponding parameter sets could be determined.

Some... ...synthesizing a wideband signal using an LPC synthesis approach followed by highpass and spectral shaping **filters**. The method according to the present invention also belongs to this category of parametric with... ...converted first to LP parameters. These LP parameters are then used to construct a synthesis **filter**, which needs to be excited by a suitable wideband excitation signal.

Two alternative approaches, commonly... ...and 5B. First, as shown in Fig. 5A, the narrowband input speech signal is inverse **filtered** 72 using previously extracted LP coefficients to obtain a narrowband residual signal. This is accomplished... ...flattening can be done by applying an LPC analysis to this signal, followed by inverse **filtering**.

A second and preferred alternative is shown in Fig. 5B. It is useful for reducing... ...need to perform the necessary additional interpolation in the first scheme. To perform the inverse **filtering** 84, the option exists in this case for either using the wideband LP parameters obtained from the mapping stage to get the inverse **filter** coefficients, or inserting zeros, like in spectral folding, into the narrowband LP coefficient vector. The... ...when a nonlinear operator is used, i.e., using the original LP coefficients for inverse **filtering** 72 the input narrowband signal followed by interpolation. The bandwidth of the resulting residual signal... ...637-655, 1971; and H. Wakita, Direct Estimation of the Vocal Tract Shape by Inverse **Filtering** of Acoustic Speech Waveform, IEEE Trans. Audio and Electroacoust., vol. AU-21, No. 5, pp... ...sections of equal length, as schematically shown in Fig. 6. Moreover, an equivalence of the

filtering process by the acoustic tube and by the LP all-pole **filter** model of the pre-emphasized speech has been shown to exist under the constraint: M... ...to compensate for the glottal wave shape and lip radiation. Typically, a fixed pre-emphasis **filter** is used, usually of the form 1 - .(micro).z -1)), where .(micro). is chosen to... ...of the original wideband speech signal from which the narrowband signal was extracted by lowpass **filtering**. Using the approach according to the present invention, one can find a refinement as demonstrated... ...byupsampling 112, for example, by inserting a zero sample following each input sample and lowpass **filtering** at 4 kHz, yielding the narrowband interpolated signal S(tilde) nb)) . The symbol "(tilde)" relates to narrowband interpolated signals. Because of the spectral folding caused by upsampling, high **energy** formants at **low frequencies**, typically present in voiced speech, are reflected to high frequencies and need to be strongly attenuated by the lowpass **filter** (not shown). Otherwise, relatively strong undesired signals may appear in the synthesized highband.

Preferably, the lowpass **filter** is designed using the simple window method for FIR **filter** design, using a window function with sufficiently high sidelobes attenuation, like the Blackman window. See... ...increases with frequency, as

desired here. The frequency response of a 129 long FIR lowpass **filter** designed with a Blackman window and used in simulations is shown in Fig. 9.

In... ...frame update is used. The signal is first pre-emphasized using a first order FIR **filter** 1- .(micro).z -1)), with .(micro). = rho1)), where, as mentioned above, rho1)) is the correlation... ...in this paragraph are all performed by the LPC analysis module 114. The corresponding inverse **filter** transfer function is then given by A nb)) (z) : A nb z = 1 + Sigma i... ...kHz if f s nb = 8 kHz), the interpolated signal S(tilde) nb)) is inverse **filtered** by A nb)) (z 2))), as shown by block 126. The **filter** coefficients, which are denoted by a nb)) (down/uparrow)2, are simply obtained from a... ...i.e., inserting zeros - as done for spectral folding. Thus, the coefficients of the inverse **filter** A nb)) (z 2))), operating at the high sampling frequency, including the unity leading term... ...not to be confused with A nb)) (z) in equ. (3), which denotes the inverse-**filter** transfer function, are computed 116 from the partial correlation coefficients (parcors) of the narrowband signal... ...coefficients represent a wideband spectral envelope.

To synthesize the highband signal, the wideband LPC synthesis **filter** 122, which uses these coefficients, needs to be excited by a signal that has energy... ...the whole upper band, is a desired feature and eliminates the need to apply any **filtering** in addition to highpass **filtering** 134. Fullwave rectification is preferred. A memoryless nonlinearity maintains signal periodicity, thus avoiding artifacts caused... ...present invention also takes into account that the highband signal of natural wideband speech has **pitch** dependent time-envelope **modulation**, which is preserved by the nonlinearity. The inventor's preference of fullwave rectification over the... ...of spectral tilt is desired, then either the wideband excitation can be flattened via inverse **filtering**, as discussed above, or infinite clipping can be used having the characteristics shown in Fig... ...is not identical to the original input narrowband signal, the synthesized signal is preferably highpass **filtered** 134 and the resulting highband signal, S hb)), is gain adjusted 134 and added 136 ... ...out put signal S(circumflex).wb)). Note that like the gain factor, also the highpass **filter** can be applied either before or after the wideband LPC synthesis block.

While Fig. 8...done in Fig. 8 (the HPF needs then to be replaced by a proper shaping **filter** to attenuate high frequencies, as discussed earlier). The use of spectral folding is, of course... ...above residual signal r(tilde) nb)) (i.e., obtained by using a wb))), but highpass **filter** its output, and combine it (after proper gain adjustment) with the interpolated narrowband residual signal... ...wideband excitation signal r wb)). This signal is fed then into the wideband LPC synthesis **filter**. Here again the resulting signal, y wb)), can serve as the desired output signal.

Various... ...narrowband module from Fig. 8 may comprise the 1:2 interpolation block 112, the inverse **filter** 126 and the elements 128, 130 and 132 to generate an excitation signal from the... ...signal.

Another way to generate a highband signal is to excite the wideband LPC synthesis **filter** (constructed from the wideband LPC coefficients) by white noise and apply highpass **filtering** to the synthesized signal. While this is a well-known simple technique, it suffers from... ...Fig. 9 illustrates a graph 138 includes the frequency response of a low pass interpolation **filter** used for 2:1 signal interpolation. Preferably, the **filter** is a half-band linear-phase FIR **filter**, designed by the window method using a Blackman window.

When the narrowband speech is obtained... ...sector of the International Telecommunication Union), for analog telephone channels. The frequency response of a **filter** that simulates the IRS characteristics is shown in Fig. 10 as a dashed line 146... ...mitigate them. Also shown in Fig. 10 are the frequency response associated with a compensation **filter** 142 and the response associated with the cascade of the two (compensated response).

One aspect... ...to the above, and independently of it, it is useful to use an extended highpass **filter**, having a cutoff frequency F c)) matched to the upper edge of the signal band... ...addition to an IRS channel response 146, Fig. 10 shows the response of a compensating **filter** 142 and the resulting compensated response 144, which is flat in the nominal range. The compensation **filter** designed here is an FIR **filter** of length 129. This number could be lowered even to 65, with only little effect. The compensated signal becomes then the input to the bandwidth extension system. This **filtering** of the output signal from a telephone channel would then be added as a block... ...art, the lowerband signal may be generated by just applying a narrow (300 Hz) lowpass **filter** to the synthesized wideband signal in parallel to the highpass **filter** 134 in Fig. 8. Other known work in the art addresses this issue more carefully... ...H. Yasukawa, Restoration of Wide Band Signal from Telephone Speech using Linear Prediction Residual Error **Filtering**, in Proc. IEEE Digital Signal Processing Workshop, pp. 176-178, 1996. This approach includes adding to the proposed system a 300 Hz LPF in parallel to the existing highpass **filter**. However, because the nonlinear operator injects also undesired components into the lowband (as excitation), audible... ...signal, S nb)), by a factor, such as a factor of 2 (upsampling and lowpass **filtering**). This step results in a narrowband interpolated signal S(tilde) nb)) . The signal S(tilde) nb)) is inverse **filtered** (166) using, for example, a transfer function of A nb)) (z 2))) having the coefficients... ...rate.

Next, a non-linear operation is applied to the signal output from the inverse **filter**. The operation comprises fullwave rectification (absolute value) of residual signal S(tilde) nb)) (168). Other... ...to signal rectification (as discussed

below) via LPC analysis of the rectified signal and inverse **filtering**. The preferred setting here is no spectral tilt compensation.

Next, the highband signal must be... ...added (174) to the original narrowband signal. This step comprises exciting a wideband LPC synthesis **filter** (170) (with coefficients a wb))) by the generated wideband excitation signal r wb)), resulting in... ...undergo further processing. If further processing is desired, the wideband signal y wb)) is highpass **filtered** (172) using a HPF having its cutoff frequency at F c)) to generate a highband... ...original highband signal, which is maintained also in the generated highband signal.

Applying a dispersion **filter** such as an allpass nonlinear-phase **filter**, as in the 2400 bps DoD standard MELP coder, for example, can mitigate the spiky... ...i wb and the wideband excitation signal, generating a highband signal S hb)) by highpass **filtering** y wb)), adjusting the gain and generating the wideband signal by summing the synthesized highband... ...a signal obtained bypassing a white Gaussian signal, v(n), through a half-band lowpass **filter** are discussed followed by some specific nonlinear memoryless operators, namely-generalized rectification, defined below, and... ...McGraw-Hill, New York, 1965 ("Papoulis").

Referring to Fig. 18, the signalv(n) is lowpass **filtered** 320 to produce x(n) and then passed through a nonlinear operator 322 to produce a signal z(n). The lowpass **filtered** signal x(n) has, ideally, a flat spectral magnitude for -pi / 2 <= theta <= pi / 2... ...n) has zero mean and variance sigma v 2 , and that the half-band lowpass **filter** is ideal, the autocorrelation functions of v(n) and x(n) are: R v m...in mitigating the 'spectral gap' near 4 kHz. It also helps when a narrow lowpass **filter** is used to extract from the synthesized wideband signal a synthetic lowband (0 - 300 Hz... ...to be useful. It can be added to the bandwidth extension system as a preprocessing **filter** at its input, as demonstrated herein.

It should be noted that when the input signal...

Claims: ...wideband signal from a narrowband signal of claim 16, the method further comprising:

(8) highpass **filtering** the wideband signal ywb)) to generate a highband signal; and

(9) combining the highband signal ... ...the bandwidth of a narrowband signal of claim 21, the method further comprising:

(7) highpass **filtering** the wideband signal ywb)) to produce a highband signal; and

(8) combining the highband signal ... ...wideband LPCs; and

(5) synthesizing a wideband signal ywb)) using the wideband LPCs and highpass

filtered white noise in the higher band of an excitation signal and a linear prediction residual... ...24, wherein computing the excitation signal from a narrowband prediction residual signal further comprises inverse **filtering** the narrowband signal.

26. A method of producing a wideband signal from a narrowband signal... ...wideband signal ywb)) from the wideband LPCs awbi and the wideband excitation signal;

(8) highpass **filtering** the wideband signal ywb)) to produce a highband signal; and

(9) generating a wideband signal... ...narrowband signal to produce an upsampled narrowband signal;

producing a narrowband residual signal by inverse **filtering** the upsampled interpolated narrowband signal using a transfer function associated with the awbiLP coefficients; and... ...wideband signal from a narrowband signal of claim 28, the method further comprising:

(10) highpass **filtering** the wideband signal ywb)) to generate a highband signal Shb)); and

(11) generating a wideband ... ...signal to produce an upsampled interpolated narrowband signal;

producing a narrowband residual signal by inverse **filtering** the upsampled interpolated narrowband signal using a transfer function associated with the awbi LP coefficients...

Claims: ...excitation signal.

14. A method as claimed in claim 13, the method further comprising:highpass **filtering** the wideband signal y wb)) to generate a highband signal; and

combining the highband signal ... ...excitation signal.

19. A method as claimed in claim 18, the method further comprising:highpass **filtering** the wideband signal y wb)) to produce a highband signal; and

combining the highband signal ... ...wideband LPCs; and

synthesizing the wideband signal y wb)) using the wideband LPCs and highpass **filtered** white noise in the higher band of an excitation signal and a linear prediction residual... ...21, wherein computing the excitation signal from a narrowband prediction residual signal further comprises inverse **filtering** the narrowband signal.

23. A method as claimed in claim 2, wherein the step of... ...y wb)) from the wideband LPCs a i wb and

the wideband excitation signal;

highpass **filtering** the wideband signal y wb)) to produce a highband signal; and

generating a wideband signal... ...produce an upsampled narrowband signal;

producing a narrowband residual signal r(tilde) nb)) by inverse **filtering** the upsampled interpolated narrowband signal using a transfer function associated with the a iwb LP... ...excitation signal.

26. A method as claimed in claim 25, the method further comprising:highpass **filtering** the wideband signal y wb)) to generate a highband signal S hb)); and

generating a... ...an upsampled interpolated narrowband signal;

producing a narrowband residual signa r(tilde) nb)) by inverse **filtering** the upsampled interpolated narrowband signal using a transfer function associated with the a i wb...

Claims: ...nach Anspruch 21, worin das Berechnen des Anregungssignals aus einem schmalbandigen Pradiktions-Restsignal ausserdem inverse **Filterung** des Schmalbandsignals umfasst.

23. Verfahren nach Anspruch 2, worin der Schritt des Berechnens der M... ...um ein aufwartsgetastetes Schmalbandsignal zu erzeugen;

Erzeugen eines Schmalband-Restsignals r(tilde) nb)) durch inverse **Filterung** des aufwartsgetasteten interpolierten Schmalbandsignals unter Verwendung einer mit den a i wb linearen Pradiktions-Koeffizienten... ...ein aufwartsgetastetes interpoliertes Schmalbandsignal zu erzeugen;

Erzeugen eines Schmalband-Restsignals r(tilde) nb)) durch inverse **Filterung** des aufwartsgetasteten interpolierten Schmalbandsignals unter Verwendung einer mit den a i wb linearen Pradiktions-Koeffizienten...

13/3K/2 (Item 2 from file: 348) Links
EUROPEAN PATENTS

01179408
Coding apparatus

Kodiergerat
Dispositif de codage

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International Patent Class (V7): G10L-019/06; G10L-019/00; G10L-019/06... ...G10L-019/00Abstract ...The coding apparatus comprises an adaptive codebook (4) storing excitation signals as vectors, a synthesis **filter** (15) for forming a synthesis signal, referring to the vectors stored in the adaptive codebook... ...computation circuit (16) for computing a similarity between the synthesis signal obtained by the synthesis **filter** and a target signal, and a coding scheme determining circuit (17) for deciding one coding...

Abstract Word Count: 93
NOTE: 1
NOTE: Figure number on first page: 1

Type	Pub. Date	Kind	Text
Publication: English			
Procedural: English			
Application: English			

Available Text	Language	Update	Word Count
CLAIMS A	(English)	200033	347
SPEC A	(English)	200033	15789
CLAIMS B	(English)	200332	334
CLAIMS B	(German)	200332	298

CLAIMS B	(French)	200332	396
SPEC B	(English)	200332	15856
Total Word Count (Document A) 16138			
Total Word Count (Document B) 16884			
Total Word Count (All Documents) 33022			

Specification: ...selected on the basis of the weighting sum value of seven characteristic amounts including a **low frequency** speech **energy**, a zero-cross ratio, and the likes.

Although the coding system select methods as described... ...limited even if recovery processing of pitch information is performed with use of a post **filter** in the decoding side.

Further, if coded data transferred with a ...In the present invention, a reference vector is extracted from an adaptive codebook and is **filtered** by the synthesize **filter** from which a synthesize signal is generated, and the similarity between the synthesize signal and... ...easily attained even if the bit number assigned to a drive signal of the synthetic **filter** is reduced. In brief, the coding bit rate can be lowered. Inversely, when a target... ...in case where the calculation amount for deciding a reference vector inputted into a synthesize **filter** is relatively large, and the calculation amount for selecting a coding scheme is remarkably small... an output terminal 13. The selection section 11 comprises an adaptive codebook 14, a synthesis **filter** 15, a similarity calculator 106, and a coding scheme determining circuit 17.

In the next... ...signal q(n) is generated from the vector p(n), by means of a synthesis **filter** 105. As a example, operation of the synthesis **filter** 105 can be expressed by the following equation (1) with respect to a z-conversion... ...a step S11, and then, the vector p(n) is made pass through a synthesis **filter** 105, to prepare a synthesis vector q(n). Next, an optimum gain g to be...and outputs T which minimizes the power E of a prediction error signal of the **prediction**, as a **pitch** period. Specifically, the **prediction** error signal power E is expressed as follows. Here, g denotes a pitch gain and... ...n) in case where the target signal r(n) is weighted by a hearing weighting **filter**. In addition, since envelope information 0 of a speech signal can be removed with use... ...v(n) obtained by making an input speech signal u(n) pass through an LPC **prediction filter**, much excellent **pitch** analysis can be achieved. Accordingly, in this embodiment, an input speech signal u(n) or... ...Further, in this embodiment, although explanation has been made to a case where a primary **pitch prediction filter** is used in the **pitch** analyzer 22, a **prediction filter** of a higher order may be used.

FIG. 5 is a block diagram showing the ... ...candidates, synthesis vectors are respectively obtained with respect to the reference vectors by the synthesis **filter** 15, and the synthesis vector most similar to the target vector r(n) is searched... an adaptive codebook 14, and coding scheme selection information I is outputted through a synthesis **filter** 15, a similarity calculator 16, and a coding scheme selection section 17 on the basis of the reference vector p(n). The processing performed by the synthesis **filter** 15, the similarity calculator 16, and the coding scheme selection section 17 are respectively the... ...signal may be of a signal which has been made pass through a hearing weighting **filter** and on which influences from a previous frame has been reduced, in several cases. Those...which will be referred to as an LPC coefficient, hereinafter) is obtained thereby. A synthesis **filter** 63 whose characteristic is defined by the LPC coefficient is inputted with an adaptive vector... ...each other by an adder 73, thereby to generate a drive signal for a synthesis **filter** 75.

Meanwhile, the characteristic of the synthesis **filter** 75 is defined on the basis of an LPC coefficient obtained by quantizing an LPC... ...74, and a drive signal outputted from an adder 73 is inputted into the synthesis **filter** 75, thereby generating a synthesis signal. With a signal from which influences of a previous... ...frame, to obtain an error signal.

The error signal is weighted by a hearing weighting **filter** 78, and thereafter, the electric power of the signal is obtained by an error calculator... ...71 after multiplication by a pitch gain, thereby generating a drive signal for a synthesis **filter** 83. In the next, an LPC coefficient is quantized by an LPC quantizer 82, and the characteristic of a synthesis **filter** 83 is defined on the basis of the LPC coefficient after the quantization. The synthesis **filter** 83 is inputted with a drive signal outputted from the multiplier 89, and a synthesis... ...an error signal is thereby obtained.

The error signal is weighted by a hearing weighting **filter** 85, and thereafter, the electric power is obtained by an error calculator 86. A gain... ...codebook 67 as a component of an encoder of the CELP method and a synthesis **filter** 63 are used for selection of an encoder (or coding scheme), and therefore, it is... ...Therefore, even if the number of bits assigned to a drive signal for the synthesis **filter** is reduced to be small, it is possible to easily attain target quality and to... ...by making a reference vector obtained from the adaptive codebook 67 pass through the synthesis **filter** 73 and an input speech signal as a target signal is obtained by a similarity...method of obtaining a pitch period and a pitch gain with use of a primary **pitch prediction filter**, a higher order prediction **filter** may be used. In addition, another pitch analysis method, e.g., a zero-crossing method... ...n). Here, explanation will be made to a case of using an all-pole pitch **filter**. The transmit function of a pole type pitch **filter** can be expressed as follows. Here, A(z) denotes a z-transformation value of an... ...smaller than 1, and (epsilon) = 0.8 is recommended. To avoid making of an oscillation **filter**, it is necessary to monitor such that a product of g and (epsilon) is always... ...The above explanation has been made to a case of using a primary pitch emphasis **filter**. However, the number of stages of the pitch emphasis **filter**

must not always be one stage, but the pitch emphasis **filter** may be stages equal in number to the number of analysis stages of the pitch... ...although the above explanation has been made to a case where a pole type pitch **filter** is used, it is naturally possible to use, for example, an all-zero pitch **filter**, pole-zero pitch **filter** , etc.

Although the characteristic is changed depending on the pitch gain g in the pitch...emphasis section 100 shown in FIG. 39 has a structure obtained by adding a prediction **filter** 104 supplied with an input signal, a LPC analyzer 105 and a synthesis **filter** 106 to the emphasis section shown in FIG. 12. The contents of the processing will... ...the like, and any of these methods can be used. In the next, a prediction **filter** is formed from an LPC coefficient, and an input signal is made pass through the prediction **filter**, thereby to generate a prediction remaining difference signal d(n) (in a step S1102). The... ...with a(n) of the equation (16) replaced with d(n).

At last, a synthesis **filter** is formed from an LPC coefficient, and the pitch emphasis signal b(n) is made pass through the synthesis **filter** to generate a pitch-emphasized input signal e(n) (in a step S1105). where n... ...an index (number) are extracted. The LPC coefficient ai is supplied to an LPC synthesis **filter** 213. Note that P is a prediction number of stages and P = 10 is generally used. A transmit function for an LPC synthesis **filter** 213 is supplied by the following equation (23).

In the next, explanation will be made...At first, an influence onto a current frame from an internal state of the synthesis **filter** 213 in a previous frame is subtracted from one frame of speech signals inputted into... ...the sub-frames.

A drive signal vector as an input signal of an LPC synthesis **filter** 213 is obtained by adding a value, which is obtained by multiplying an adaptive vector... ...a multiplier 210, by means of an adder 212.

Here, the adaptive codebook 207 performs **pitch prediction** analysis described in the prior art reference 1, through closed loop operation or analysis by... ...art reference 2). According to the reference 2, a drive signal for the LPC synthesis **filter** 213 is delayed by one sample by a delay circuit 211 for a pitch search... ...one after another, and are respectively multiplied by predetermined gains obtained from the multiplier 209. **Filter** processing is performed by an LPC synthesis **filter** 213, and a synthesis signal vector is generated. The synthesis signal vector thus generated is... ...a subtracter 203. An output of the subtracter 203 is inputted through a hearing weighting **filter** 204 to an error calculator 205, and an average quadratic error is obtained. Information concerning... ...steps is multiplied by a gain, and a synthesis speech signal vector is generated through **filter** calculation by the LPC synthesis **filter** 213. The vector thus generated is subtracted from a target vector, thereby resulting in a... ...multiplication by a gain obtained from the gain codebook 218 by the multiplier 210, to **filter** calculation by the LPC synthesis **filter** 213. Thereafter, generation of a synthesis speech signal vector and calculation of an average square... ...and 210 are each transmitted from an index selector 214. Note that the hearing weighting **filter** 204 is used to shape a spectrum of an error signal outputted from a subtracter... ...adder 403 to generate a drive vector which is made pass through an LPC synthesis **filter** 404 whose setting is performed by an LPC coefficient transmitted from a coding section, thereby... ...subjective quality of the synthesis signal, the synthesis signal is made pass through a post **filter** 405 to obtain a synthesis speech which is outputted through an output terminal 406. Finally...

Specification: ...selected on the basis of the weighting sum value of seven characteristic amounts including a

low frequency speech **energy**, a zero-cross ratio, and the likes.

Although the coding system select methods as described...limited even if recovery processing of pitch information is performed with use of a post **filter** in the decoding side.

Further, if coded data transferred with a transfer path code added... ...In the present invention, a reference vector is extracted from an adaptive codebook and is **filtered** by the synthesize **filter** from which a synthesize signal is generated, and the similarity between the synthesize signal and... ...easily attained even if the bit number assigned to a drive signal of the synthetic **filter** is reduced. In brief, the coding bit rate can be lowered. Inversely, when a target... ...in case where the calculation amount for deciding a reference vector inputted into a synthesize **filter** is relatively large, and the calculation amount for selecting a coding scheme is remarkably small... an output terminal 13. The selection section 11 comprises an adaptive codebook 14, a synthesis **filter** 15, a similarity calculator 106, and a coding scheme determining circuit 17.

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input speech signal u(n) or... ...Further, in this embodiment, although explanation has been made to a case where a primary **pitch prediction filter** is used in the **pitch** analyzer 22, a **prediction filter** of a higher order may be used.

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Meanwhile, the characteristic of the synthesis **filter** 75 is defined on the basis of an LPC coefficient obtained by quantizing an LPC... ...74, and a drive signal outputted from an adder 73 is inputted into the synthesis **filter** 75, thereby generating a synthesis signal. With a signal from which influences of a previous... ...frame, to obtain an error signal.

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The error signal is weighted by a hearing weighting **filter** 85, and thereafter, the electric power is obtained by an error calculator 86. A gain... ...codebook 67 as a component of an encoder of the CELP method and a synthesis **filter** 63 are used for selection of an encoder (or coding scheme), and therefore, it is... ...Therefore, even if the number of bits assigned to a drive signal for the synthesis **filter** is reduced to be small, it is possible to easily attain target quality and to... ...by making a reference vector obtained from the adaptive codebook 67 pass through the synthesis **filter** 73 and an input speech signal as a target signal is obtained by a similarity...method of obtaining a pitch period and a pitch gain with use of a primary **pitch prediction filter**, a higher order prediction **filter** may be used. In addition, another pitch analysis method, e.g., a zero-crossing method... ...n). Here, explanation will be made to a case of using an all-pole pitch **filter**. The transmit function of a pole type pitch **filter** can be expressed as follows. Here, A(z) denotes a z-transformation value of an... ...smaller than 1, and (epsilon) = 0.8 is recommended. To avoid making of an oscillation **filter**, it is necessary to monitor such that a product of g and (epsilon) is always... ...The above explanation has been made to a case of using a primary pitch emphasis **filter**. However, the number of stages of the pitch emphasis **filter** must not always be one stage, but the pitch emphasis **filter** may be stages equal in number to the number of analysis stages of the pitch... ...although the above explanation has been made to a case where a pole type pitch **filter** is used, it is naturally possible to use, for example, an all-zero pitch **filter**, pole-zero pitch **filter** , etc.

Although the characteristic is changed depending on the pitch gain g in the pitch...emphasis section 100 shown in FIG. 39 has a structure obtained by adding a prediction **filter** 104 supplied with an input signal, a LPC analyzer 105 and a synthesis **filter** 106 to the emphasis section shown in FIG. 12. The contents of the processing will... ...the like, and any of these methods can be used. In the next, a prediction **filter** is formed from an LPC coefficient, and an input signal is made pass through the prediction **filter**, thereby to generate a prediction remaining difference signal d(n) (in a step S1102). The... ...with a(n) of the equation (16) replaced with d(n).

At last, a synthesis **filter** is formed from an LPC coefficient, and the pitch emphasis signal b(n) is made pass through the synthesis **filter** to generate a pitch-emphasized input signal e(n) (in a step S1105). where n...an index (number) are extracted. The LPC coefficient ai is supplied to an LPC synthesis **filter** 213. Note that P is a prediction number of stages and P = 10 is generally used. A transmit function for an LPC synthesis **filter** 213 is supplied by the following equation (23).

In the next, explanation will be made... ...At first, an influence onto a current frame from an internal state of the synthesis **filter** 213 in a previous frame is subtracted from one frame of speech signals inputted into... ...the sub-frames.

A drive signal vector as an input signal of an LPC synthesis **filter** 213 is obtained by adding a value, which is obtained by multiplying an adaptive vector... ...a multiplier 210, by means of an adder 212.

Here, the adaptive codebook 207 performs **pitch prediction** analysis described in the prior art reference 1, through closed loop operation or analysis by... ...art reference 2). According to the reference 2, a drive signal for the LPC synthesis **filter** 213 is delayed by one sample by a delay circuit 211 for a pitch search... ...one after another, and are respectively multiplied by predetermined gains obtained from the multiplier 209. **Filter** processing is performed by an LPC synthesis **filter** 213, and a synthesis signal vector is generated. The synthesis signal vector thus generated is... ...a subtracter 203. An output of the subtracter 203 is inputted through a hearing weighting **filter** 204 to an error calculator

205, and an average quadratic error is obtained. Information concerning... ...steps is multiplied by a gain, and a synthesis speech signal vector is generated through **filter** calculation by the LPC synthesis **filter** 213. The vector thus generated is subtracted from a target vector, thereby resulting in a... ...multiplication by a gain obtained from the gain codebook 218 by the multiplier 210, to **filter** calculation by the LPC synthesis **filter** 213. Thereafter, generation of a synthesis speech signal vector and calculation of an average square... ...and 210 are each transmitted from an index selector 214. Note that the hearing weighting **filter** 204 is used to shape a spectrum of an error signal outputted from a subtracter...adder 403 to generate a drive vector which is made pass through an LPC synthesis **filter** 404 whose setting is performed by an LPC coefficient transmitted from a coding section, thereby... ...subjective quality of the synthesis signal, the synthesis signal is made pass through a post **filter** 405 to obtain a synthesis speech which is outputted through an output terminal 406. Finally...

13/3K/3 (Item 3 from file: 348) Links
EUROPEAN PATENTS

01164202
PERIODICITY ENHANCEMENT IN DECODING WIDEBAND SIGNALS
VERBESSERUNG DER PERIODIZITAT EINES BREITBANDSIGNALS
AMELIORATION DE LA PERIODICITE DANS LE DECODAGE DE SIGNAUX A LARGE BANDE

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CLAIMS B	(English)	200331	3457
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CLAIMS B	(French)	200331	4433
SPEC B	(English)	200331	8048
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Specification: ...to a method and device for enhancing periodicity of the excitation of a signal synthesis **filter** in view of producing a synthesized wideband signal. 2. Brief description of the prior art... ...and wireless applications, as well as Internet and packet network applications. Until recently, telephone bandwidths **filtered** in the range 200-3400 Hz were mainly used in speech coding applications. However, there... ...number (corresponding to 10-30 ms of speech). In CELP, a linear prediction (LP) synthesis **filter** is computed and transmitted every frame. The L-sample frame is then divided into smaller... ...signal is transmitted and used at the decoder as the input of the LP synthesis **filter** in order to obtain the synthesized speech.

An innovative codebook in the CELP context, is... ...synthesize speech according to the CELP technique, each block of N samples is synthesized by **filtering** an appropriate codevector from a codebook through time varying **filters** modeling the spectral characteristics of the speech signal. At the encoder end, the synthesis output... ...perceptually weighted distortion measure. This perceptual weighting is performed using a so-called perceptual weighting **filter**, which is usually derived from the LP synthesis **filter**.

A known CELP-based coder is described in the document EP-A-0788091.

The CELP ... ...improves the quality in case of voiced segments. This was done in the past by **filtering** the innovative codevector from the fixed codebook through a **filter** having a transfer function of the form 1/(1-(epsilon)bz-T)) where (epsilon) is... ...invention is to propose a new alternative approach by which periodicity enhancement is achieved through **filtering** the innovative codevector by an innovation **filter** which reduces the low-frequency contents of the innovative codevector, whereby the innovative contribution is... ...in relation to a pitch codevector and an innovative codevector for supplying a signal synthesis **filter** in view synthesizing a wideband signal. In this periodicity enhancing method, a periodicity factor related to the wideband signal is calculated. Then, the innovative codevector is **filtered** in relation to the periodicity factor to thereby reduce **energy** of a **low frequency** portion of the innovative codevector and enhance periodicity of a low frequency portion of the ...excitation signal produced in relation to adaptive and innovative codevectors for supplying a signal synthesis **filter** in view of synthesizing a wideband signal, comprises:

a) a factor generator for calculating a periodicity factor related to said wideband signal; and

b) an innovative **filter** for **filtering** the innovative codevector in relation to the periodicity factor to thereby reduce **energy** of a **low frequency** portion of the innovative codevector and enhance periodicity of a low frequency portion of the excitation signal.

According to a first preferred embodiment:

- the innovative codevector is **filtered** with a transfer function of the form: where (alpha) is the periodicity factor derived from... ...of the innovative codevector.

According to a second preferred embodiment:

- the the innovative codevector is **filtered** with a transfer function of the form: where (sigma) is a periodicity factor derived from... ...from this encoded wideband signal at least pitch codebook parameters, innovative codebook parameters, and synthesis **filter** coefficients;

b) an pitch codebook responsive to the pitch codebook parameters for producing a pitch... ...factor generator for calculating a periodicity factor related to the wideband signal; and the innovation **filter** for **filtering** the innovative codevector in relation to the periodicity factor;

e) a combiner circuit for combining the pitch codevector and the innovative codevector **filtered** by the innovation **filter** to thereby produce a periodicity-enhanced excitation signal; and

f) a signal synthesis **filter** for **filtering** that periodicity-enhanced excitation signal in relation to the synthesis **filter** coefficients to thereby produce the synthesized wideband signal.

According to the present invention, in a... ...from this encoded wideband signal at least pitch codebook parameters, innovative codebook parameters, and synthesis **filter** coefficients; an pitch codebook responsive to the pitch codebook

parameters for producing a pitch codevector... ...codevector and the innovative codevector to thereby produce an excitation signal; and a signal synthesis **filter** for **filtering** that excitation signal in relation to the synthesis **filter** coefficients to thereby produce the synthesized wideband signal;

the improvement therein comprising a periodicity enhancing ... ...factor generator for calculating a periodicity factor related to the wideband signal; and the innovation **filter** for **filtering** the innovative codevector in relation to the periodicity factor before supplying this innovative codevector to...and below such as Code-Excited Linear Prediction (CELP) encoders typically use a LP synthesis **filter** to model the short-term spectral envelope of the voice signal. The LP information is... ...signal in the frame are computed, encoded, and transmitted. LP parameters representing the LP synthesis **filter** are usually computed once every frame. The frame is further divided into smaller blocks of... ...sampling, preprocessing, and preemphasis);

sw)) Weighted speech vector;

s0)) Zero-input response of weighted synthesis **filter**;

sp)) Down-sampled pre-processed signal; Oversampled synthesized speech signal;

s' Synthesis signal before deemphasis... ...x Target vector for pitch search;

x' Target vector for innovation search;

h Weighted synthesis **filter** impulse response;

vT)) Adaptive (pitch) codebook vector at delay T;

yT)) **Filtered** pitch codebook vector (vT)) convolved with h);

ck)) Innovative codevector at index k (k-th... ...codebook index);

b Pitch gain (or pitch codebook gain);

j Index of the low-pass **filter** used on the pitch codevector;

...optional pre-processing block 102. Pre-processing block 102 may consist of a high-pass **filter** with a 50 Hz cut-off frequency. High-pass **filter** 102 removes the unwanted sound components below 50 Hz.

The down-sampled pre-processed signal... ...at a sampling frequency of 12.8 kHz). In a preferred embodiment of the preemphasis **filter** 103, the signal Sp))(n) is preemphasized using a **filter** having the following transfer function: where (mu) is a preemphasis factor with a value located between 0 and 1 (a typical value is (mu) = 0.7). A higher-order **filter** could also be used. It should be pointed out that high-pass **filter** 102 and preemphasis **filter** 103 can be interchanged to obtain more efficient fixed-point implementations.

The function of the preemphasis **filter** 103 is to enhance the high frequency contents of the input signal. It also reduces... ...quality. This will be explained in more detail herein below.

The output of the preemphasis **filter** 103 is denoted s(n). This signal is used for performing LP analysis in calculator... ...are computed from the windowed signal, and Levinson-Durbin recursion is used to compute LP **filter** coefficients, ai)), where i=1,...,p, and where p is the LP order, which is... ...wideband coding. The parameters ai)) are the coefficients of the transfer function of the LP **filter**, which is given by the following relation:

LP analysis is performed in calculator module 104, which also performs the quantization and interpolation of the LP **filter** coefficients. The LP **filter** coefficients are first transformed into another equivalent domain more suitable for quantization and interpolation purposes... ...are two domains in which quantization and interpolation can be efficiently performed. The 16 LP **filter** coefficients, ai)), can be quantized in the order of 30 to 50 bits using split... ...or a combination thereof. The purpose of the interpolation is to enable updating the LP **filter** coefficients every subframe while transmitting them once every frame, which improves the encoder performance without increasing the bit rate. Quantization and interpolation of the LP **filter** coefficients is believed to be otherwise well known to those of ordinary skill in the... ...rest of the coding operations performed on a subframe basis. In the following description, the **filter** A(z) denotes the unquantized interpolated LP **filter** of the subframe, and the **filter** A(z) denotes the quantized interpolated LP **filter** of the subframe.

Perceptual Weighting:

In analysis-by-synthesis encoders, the optimum pitch and innovation... ...and weighted synthesis speech.

The weighted signal sw))(n) is computed in a perceptual weighting **filter** 105. Traditionally, the weighted signal sw))(n) is computed by a weighting **filter** having a transfer function W(z) in the form: where

As well known to those... ...W-1)(z), which is the inverse of the transfer function of the perceptual weighting **filter** 105. This result is well described by B.S. Atal and M.R. Schroeder in...is controlled by the factors (gamma)1)) and (gamma)2)).

The above traditional perceptual weighting **filter** 105 works well with telephone band signals. However, it was found that this traditional perceptual weighting **filter** 105 is not suitable for efficient perceptual weighting of wideband signals. It was also found that the traditional perceptual weighting **filter** 105 has inherent limitations in modelling the formant structure and the required spectral tilt concurrently... ...range between low and high frequencies. The prior art has suggested to add a tilt **filter** into W(z) in order to control the tilt and formant weighting of the wideband... ...solution to this problem is, in accordance with the present invention, to introduce the preemphasis **filter** 103 at the input, compute the LP

filter A(z) based on the preemphasized speech s(n), and use a modified **filter** W(z) by fixing its denominator.

LP analysis is performed in module 104 on the preemphasized signal s(n) to obtain the LP **filter** A(z). Also, a new perceptual weighting **filter** 105 with fixed denominator is used. An example of transfer function for the perceptual weighting **filter** 104 is given by the following relation: where

A higher order can be used at ... ...z) is computed based on the preemphasized speech signal s(n), the tilt of the **filter** 1/A(z/(gamma)1))) is less pronounced compared to the case when A(z... ...based on the original speech. Since deemphasis is performed at the decoder end using a **filter** having the transfer function: the quantization error spectrum is shaped by a **filter** having a transfer function W-1)(z)P-1)(z). When (gamma)2)) is set... ...which is typically the case, the spectrum of the quantization error is shaped by a **filter** whose transfer function is 1/A(z/(gamma)1))), with A(z) computed based on... ...this structure for achieving the error shaping by a combination of preemphasis and modified weighting **filtering** is very efficient for encoding wideband signals, in addition to the advantages of ease of... ...computed. This is usually done by subtracting the zero-input response s0)) of weighted synthesis **filter** W(z)/A(z) from the weighted speech signal sw)) (n). This zero-input response... ...the weighted speech vector in the subframe, and s0)) is the zero-input response of **filter** W(z)/A(z) which is the output of the combined **filter** W(z)/A(z) due to its initial states. The zero-input response calculator 108 is responsive to the quantized interpolated LP **filter** A(z) from the LP analysis, quantization and interpolation calculator 104 and to the initial states of the weighted synthesis **filter** W(z)/A(z) stored in memory module 111 to calculate the zero-input response... ...due to the initial states as determined by setting the inputs equal to zero) of **filter** W(z)/A(z). This operation is well known to those of ordinary skill in... ...the target vector x.

A N-dimensional impulse response vector h of the weighted synthesis **filter** W(z)/A(z) is computed in the impulse response generator 109 using the LP **filter** coefficients A(z) and A(z) from module 104. Again, this operation is well known... ...impulse response vector h and the open-loop pitch lag TOL)) as inputs. Traditionally, the **pitch prediction** has been represented by a pitch **filter** having the following transfer function: where b is the pitch gain and T is the... ...new sample). For pitch lags T>N, the pitch codebook is equivalent to the **filter** structure (1/(1-bz-T)), and an pitch codebook vector vT))(n) at pitch lag ...from the past excitation until the vector is completed (this is not equivalent to the **filter** structure).

In recent encoders, a higher pitch resolution is used which significantly improves the quality... ...voiced sound segments. This is achieved by oversampling the past excitation signal using polyphase interpolation **filters**. In this case, the vector vT))(n) usually corresponds to an interpolated version of the... ...minimize the mean squared weighted error E between the target vector x and the scaled **filtered** past excitation. Error E being expressed as: where yT)) is the **filtered** pitch codebook vector at pitch lag T: n=0,...,N-1.

It can be shown... ...5), which significantly simplifies the search procedure. A simple procedure is used for updating the **filtered** codevector yT)) without the need to compute the convolution for every pitch lag.

Once an... ...the search (module 107) tests the fractions around that optimum integer pitch lag.

When the **pitch predictor** is represented by a **filter** of the form 1/(1-bz-T)), which is a valid assumption for pitch lags T>N, the spectrum of the pitch **filter** exhibits a harmonic structure over the entire frequency range, with a harmonic frequency related to... ...to achieve efficient representation of the pitch contribution in voiced segments of wideband speech, the **pitch prediction filter** needs to have the flexibility of varying the amount of periodicity over the wideband spectrum... ...of wideband signals is disclosed in the present specification, whereby several forms of low pass **filters** are applied to the past excitation and the low pass **filter** with higher prediction gain is selected.

When subsample pitch resolution is used, the low pass **filters** can be incorporated into the interpolation **filters** used to obtain the higher pitch resolution. In this case, the third stage of the... ...fractions around the chosen integer pitch lag are

tested, is repeated for the several interpolation **filters** having different low-pass characteristics and the fraction and **filter** index which maximize the search criterion C are selected.

A simpler approach is to complete ... ...three stages described above to determine the optimum fractional pitch lag using only one interpolation **filter** with a certain frequency response, and select the optimum low-pass **filter** shape at the end by applying the different predetermined low-pass **filters** to the chosen pitch codebook vector vT)) and select the low-pass **filter** which minimizes the **pitch prediction** error. This approach is discussed in detail below.

Figure 3 illustrates a schematic block diagram... ...vector vT)) corresponds to the interpolated past excitation signal. In this preferred embodiment, the interpolation **filter** (in module 301, but not shown) has a low-pass **filter** characteristic removing the frequency contents above 7000 Hz.

In a preferred embodiment, K **filter** characteristics are used; these **filter** characteristics could be low-pass or band-pass **filter** characteristics. Once the optimum codevector vT)) is determined and supplied by the pitch codevector generator 302, K **filtered** versions of vT)) are computed respectively using K different frequency shaping **filters** such as 305(j)), where j=1, 2, ... , K. These **filtered** versions are denoted vf))(j)), where j=1,2, ... , K. The different vectors Vf))(j... ...response h to obtain the vectors y(j)), where j=0, 1, 2, ... , K. To **calculate** the mean squared **pitch prediction** error for each vector y(j)), the value y(j)) is multiplied by the gain **filter** 305(j)) which minimizes the mean squared **pitch prediction** error j=1, 2,...,K

To **calculate** the mean squared **pitch prediction** error e()j))) for each value of y(j)), the value y(j)) is multiplied... ...is calculated in a corresponging gain calculator 306(j)) in association with the frequency shaping **filter** at index j, using the following relationship:

In selector 309, the parameters b, T, and j are chosen based on vT)) or vf))(j)) which minimizes the mean squared **pitch prediction** error e.

Referring back to Figure 1, the pitch codebook index T is encoded and... ...approach, extra information is needed to encode the index j of the selected frequency shaping **filter** in muitiplexer 112. For example, if three **filters** are used (j=0, 1, 2, 3), then two bits are needed to represent this information. The **filter** index information j can also be encoded jointly with the pitch gain b.

Innovative codebook... ...by subtracting the LTP contribution: where b is the pitch gain and yT)) is the **filtered** pitch codebook vector (the past excitation at delay T **filtered** with the selected low pass **filter** and convolved with the inpulse response h as described with reference to Figure 3).

The... ...gain g which minimize the mean-squared error between the target vector and the scaled **filtered** codevector where H is a lower triangular convolution matrix derived from the impulse response vector... ...channel.

Memory update:

In memory module 111 (Figure 1), the states of the weighted synthesis **filter** W(z)/A(z) are updated by **filtering** the excitation signal u = gck)) + bvT)) through the weighted synthesis **filter**. After this **filtering**, the states of the **filter** are memorized and used in the next subframe as initial states for computing the zero... ...known to those of ordinary skill in the art can be used to update the **filter** states.

DECODER SIDE

The speech decoding device 200 of Figure 2 illustrates the various steps... ...scaled codevector gck)) at the output of the amplifier 224 is processed through a innovation **filter** 205.

Periodicity enhancement:

The generated scaled codevector at the output of the amplifier 224 is ... ...improves the quality in case of voiced segments. This was done in the past by **filtering** the innovation vector from the innovative codebook (fixed codebook) 218 through a **filter** in the form 1/(1-(epsilon)bz-T)) where (epsilon) is a factor below 0... ...which is part of the present invention, is disclosed whereby periodicity enhancement is achieved by **filtering** the innovative codevector ck)) from the innovative (fixed) codebook through an innovation **filter** 205 (F(z)) whose frequency response emphasizes the higher frequencies more than lower frequencies. The to derive the **filter** F(z) coefficients used in a preferred embodiment, is to relate them to the amount... ...where higher frequencies are more strongly emphasized (stronger overall slope) for higher pitch gains. Innovation **filter** 205 has the effect of lowering the energy of the innovative codevector ck)) at low ... ...the excitation signal u at lower frequencies more than higher frequencies. Suggested forms for innovation **filter** 205 are

(1) or

(2) where (sigma) or (alpha) are periodicity factors derived from the... ...pitch codevector vT)) from the pitch codebook 201 is then processed through a low-pass **filter** 202 whose cut-off frequency is adjusted by means of the index j from the... ...periodicity factor (sigma) is calculated as follows:

The enhanced signal cf)) is therefore computed by **filtering** the scaled innovative codevector gck)) through the innovation **filter** 205 (F(z)).

The enhanced excitation signal u' is computed by the adder 220 as... ...and the enhanced excitation signal u' is used at the input of the LP synthesis **filter** 206.

Synthesis and deemphasis

The synthesized signal s' is computed by **filtering** the enhanced excitation signal u' through the LP synthesis **filter** 206 which has the form 1/A(z), where A(z) is the interpolated LP **filter** in the current subframe. As can be seen in Figure 2, the quantized LP coefficients A(z) on line 225 from demultiplexer 217 are supplied to the LP synthesis **filter** 206 to adjust the parameters of the LP synthesis **filter** 206 accordingly. The deemphasis **filter** 207 is the inverse of the preemphasis **filter** 103 of Figure 1. The transfer function of the deemphasis **filter** 207 is given by where (mu) is a preemphasis factor with a value located between 0 and 1 (a typical value is (mu) = 0.7). A higher-order **filter** could also be used.

The vectors' is **filtered** through the deemphasis **filter** D(z) (module 207) to obtain the vector sd)), which is passed through the high-pass **filter** 208 to remove the unwanted frequencies below 50 Hz and further obtain sh)).

Oversampling and it with the same LP synthesis **filter** used for synthesizing the down-sampled signal S(sup AND).

The high frequency generation procedure... ...speech domain using the spectral shaper 215. In the preferred embodiment, this is achieved by **filtering** the noise wg)) through a bandwidth expanded version of the same LP synthesis **filter** used in the down-sampled domain (1/A(z/0.8)). The corresponding bandwidth expanded LP **filter** coefficients are calculated in spectral shaper 215.

The **filtered** scaled noise sequence wf)) is then band-pass **filtered** to the required frequency range to be restored using the band-pass **filter** 216. In the preferred embodiment, the band-pass **filter** 216 restricts the noise sequence to the frequency range 5.6-7.2 kHz. The resulting band-pass **filtered** noise sequence z is added in adder 221 to the oversampled synthesized speech signal s...

Claims: ...in relation to a pitch codevector and an innovative codevector for supplying a signal synthesis **filter** in view of synthesizing a wideband signal, said periodicity enhancing device comprising:

a) a factor... ...204) for calculating a periodicity factor related to the wideband signal; and

b) an innovation **filter** (205) for **filtering** the innovative codevector in relation to said periodicity factor to thereby reduce **energy** of a **low frequency** portion of the innovative codevector and enhance periodicity of a low frequency portion of the... ...innovative codevector.

3. A periodicity enhancing device as defined in claim 1, wherein said innovation **filter** has a transfer function of the form: where (alpha) is a periodicity factor derived from... ...innovative codevector.

7. A periodicity enhancing device as defined in claim 1, wherein said innovation **filter** has a transfer function of the form: where (sigma) is a periodicity factor derived from... ...in relation to a pitch codevector and an innovative codevector for supplying a signal synthesis **filter** in view of synthesizing a wideband signal, said periodicity enhancing method comprising the steps of:

a) calculating a periodicity factor related to the wideband signal; and

b) **filtering** the innovative codevector in relation to said periodicity factor to thereby reduce **energy** of a **low frequency** portion of the innovative codevector and enhance periodicity of a low frequency portion of the... ...innovative codevector.

13. A method for enhancing periodicity as defined in claim 10, wherein said **filtering** comprises processing the innovation vector through an innovation **filter** having a transfer function of the form: where (alpha) is a periodicity factor derived from... ...innovative codevector.

17. A method for enhancing periodicity as defined in claim 11, wherein said **filtering** comprises processing the innovation vector through an innovation **filter** having a transfer function of the form: where (sigma) is a periodicity factor derived from pitch codebook parameters, innovative codebook parameters, and synthesis **filter** coefficients;

b) an pitch codebook responsive to said pitch codebook parameters for producing a pitch... ...factor generator for calculating a periodicity factor related to the wideband signal, and said innovation **filter** for **filtering** the innovative codevector;

e) a combiner circuit for combining said pitch codevector and said innovative codevector **filtered** by said innovation **filter** to thereby produce said periodicity enhanced excitation signal; and

f) a signal synthesis **filter** for **filtering** said periodicity enhanced excitation signal in relation to said synthesis **filter** coefficients to thereby produce said synthesized wideband signal.

22. A decoder for producing a synthesized... ...decoder for producing a synthesized wideband signal as defined in claim 21, wherein said innovation **filter** has a transfer function of the form: where (alpha) is a periodicity factor derived from... ...decoder for producing a synthesized wideband signal as defined in claim 21, wherein said innovation **filter** has a transfer function of the form: where (sigma) is a periodicity factor derived from... ...from said encoded wideband signal at least pitch codebook parameters, innovative codebook parameters, and synthesis **filter** coefficients;

b) a pitch codebook responsive to said pitch codebook parameters for producing a pitch... ...codevector and innovative codevector to thereby produce an excitation signal; and

e) a signal synthesis **filter** for **filtering** said excitation signal in relation to said synthesis **filter** coefficients to thereby produce said synthesized wideband signal; the decoder further comprising a periodicity enhancing... ...factor generator for calculating a periodicity factor related to the wideband signal, and said innovation **filter** for **filtering** the innovative codevector.

32. A decoder for producing a synthesized wideband signal as defined in... ...decoder for producing a synthesized wideband signal as defined in claim 31, wherein said innovation **filter** has a transfer function of the form: where (alpha) is a periodicity factor derived from... ...decoder for producing a synthesized wideband signal as defined in claim 31, wherein said innovation **filter** has a transfer function of the form: where (sigma) is a periodicity factor derived from... innovative codevector.

43. A cellular communication system as defined in claim 41, wherein said innovation **filter** has a transfer function of the form: where (alpha) is a periodicity factor derived from... ...innovative codevector.

47. A cellular communication system as defined in claim 41, wherein said innovation **filter** has a transfer function of the form: where (sigma) is a periodicity factor derived from... ...53. A cellular mobile transmitter/receiver unit as defined in claim 51, wherein said innovation **filter** has a transfer function of the form: where (alpha) is a periodicity factor derived from... ...57. A cellular mobile transmitter/receiver unit as defined in claim 51, wherein said innovation **filter** has a transfer function of the form: where (sigma) is a periodicity factor derived from...innovative codevector.

63. A cellular network element as defined in claim 61, wherein said innovation **filter** has a transfer function of the form: where (alpha) is a periodicity factor derived from... ...innovative codevector.

67. A cellular network element as defined in claim 61, wherein said innovation **filter** has a transfer function of the form: where (sigma) is a periodicity factor derived from... ...73. A bidirectional wireless communication sub-system as defined in claim 71, wherein said innovation **filter** has a transfer function of the form: where (alpha) is a periodicity factor derived from... ...77. A bidirectional wireless communication sub-system as defined in claim 71, wherein said innovation **filter** has a transfer function of the form: where (sigma) is a periodicity factor derived from...

Claims: ...eines Periodizitatsfaktors, der mit dem Breitbandsignal in Beziehung steht; und

b) ein Innovationsfilter (205) zum **Filtern** des innovativen Codevektors in bezug auf den Periodizitatsfaktor, um dadurch die Energie eines niederfrequenten Abschnitts... ...Schritte umfast:

a) Berechnen eines Periodizitatsfaktors, der mit dem Breitbandsignal in Beziehung steht; und

b) **Filtern** des innovativen Codevektors in bezug auf den Periodizitatsfaktor, um dadurch die Energie eines niederfrequenten Abschnitts... ...innovativen Codevektor umfast.

13. Verfahren zum Verbessern der Periodizitat nach Anspruch 11, bei dem die **Filterung** das Verarbeiten des Innovationsvektors durch ein Innovationsfilter umfast, das eine Ubertragungsfunktion der folgenden Form besitzt... ...innovativen Codevektors ist.

17. Verfahren zum Verbessern der Periodizitat nach Anspruch 11, bei dem die **Filterung** die Verarbeitung des Innovationsvektors durch ein Innovationsfilter umfast, das eine Ubertragungsfunktion der folgenden Form besitzt... ...Faktorgenerator zum Berechnen eines mit dem Breitbandsignal in Beziehung stehenden Periodizitatsfaktors und das Innovationsfilter zum **Filtern** des innovativen Codevektors umfast;

e) eine Kombinationsschaltung zum Kombinieren des Tonhohen-Codevektors und des innovativen Signalsynthetisierungsfilter zum **Filtern** des Erregungssignals mit verbesserter Periodizitat in bezug auf die Synthetisierungsfilter-Koeffizienten, um dadurch das synthetisierte... ...und des innovativen Codevektors, um dadurch ein Erregungssignal zu erzeugen; und

e) ein Signalsynthetisierungsfilter zum **Filtern** des Erregungssignals in bezug auf die Synthetisierungsfilter-Koeffizienten, um dadurch das synthetisierte Breitbandsignal zu erzeugen... ...die einen Faktorgenerator zum Berechnen eines auf das Breitbandsignal bezogenen Periodizitatfaktors und das Innovationsfilter zum **Filtern** des innovativen Codevektors umfast.

32. Decodierer zum Erzeugen eines synthetisierten Breitbandsignals nach Anspruch 31, bei ...

13/3K/4 (Item 4 from file: 348) Links
EUROPEAN PATENTS

00785916
Coding apparatus
Kodiergerat
Dispositif de codage

Patent Assignee:

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	Country	Number	Kind	Date	
Patent	EP	734014	A1	19960925	(Basic)
	EP	734014	B1	20001115	
Application	EP	95306943		19950929	
Priorities	JP	9563660		19950323	

Designated States:
DE; FR; GB; IT;

Related Divisions: Patent (Application):EP 1028411 (EP 109508)

International Patent Class (V7): G10L-019/08; G10L-019/08Abstract ...The coding apparatus comprises an adaptive codebook (4) storing excitation signals as vectors, a synthesis **filter** (15) for forming a synthesis signal, referring to the vectors stored in the adaptive codebook... ...computation circuit (16) for computing a similarity between

the synthesis signal obtained by the synthesis **filter** and a target signal, and a coding scheme determining circuit (17) for deciding one coding...

Abstract Word Count: 116
NOTE: 1
NOTE: Figure number on first page: 1

Type	Pub. Date	Kind	Text

Publication: English
Procedural: English
Application: English

Available Text	Language	Update	Word Count
CLAIMS B	(English)	200046	369
CLAIMS B	(German)	200046	324
CLAIMS B	(French)	200046	453
SPEC B	(English)	200046	15871
Total Word Count (Document A) 0			
Total Word Count (Document B) 17017			
Total Word Count (All Documents) 17017			

13/3K/5 (Item 5 from file: 348) Links
EUROPEAN PATENTS

00228350
Improved voice coding process and device for implementing said process.
Sprachkodierungsverfahren und Einrichtung zur Ausfuhrung dieses Verfahrens.
Procede de codage de la parole et dispositif pour la mise en oeuvre dudit procede.

Patent Assignee:

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	Country	Number	Kind	Date	
Patent	EP	243562	A1	19871104	(Basic)
	EP	243562	B1	19920129	
Application	EP	86430014		19860430	
Priorities	EP	86430014		19860430	

Designated States:
DE; FR; GB; IT;

International Patent Class (V7): G10L-009/14; ; G10L-009/14Abstract ...added to the phase band prior to be used for driving a linear prediction synthesis **filter** tuned using said linear prediction parameters.

Abstract Word Count: 90

Type	Pub. Date	Kind	Text

Publication: English
Procedural: English
Application: English

Available Text	Language	Update	Word Count
CLAIMS B	(English)	EPBBF1	1299
CLAIMS B	(German)	EPBBF1	1284
CLAIMS B	(French)	EPBBF1	1578
SPEC B	(English)	EPBBF1	3754
Total Word Count (Document A) 0			
Total Word Count (Document B) 7915			
Total Word Count (All Documents) 7915			

Specification: ...analysis is made to generate, besides the low frequency bandwidth signal, parameters relating to the **high frequency** bandwidth **energy** contents and to the original voice signal spectral characteristics.

RELP methods enable reproducing speech signal... ...and parameters characterizing the high frequency bandwidth components of said voice signal said parameters including **energy indications** about said high frequency bandwidth signal, said voice coding process being further characterized in that... ...and a synthesizer. In the analyzer the input speech signal is processed to derive therefrom **the** following set of speech descriptors:

(I) the spectral descriptors represented by a set of linear prediction parameters. (see LP **Analysis** in Fig.1).

(II) the base-band signal obtained by band limiting (300-1000 Hz) and subsequently sub-sampling at 2kHz the residual (or excitation) signal resulting from **the** inverse **filtering** of the speech signal by its predictor (see BB Extraction in Fig.1) or by a conventional low frequency **filtering** operation.

(III) the energy of the upper band (or High-Frequency band) signal (1000 to 3400 Hz) which has been removed from the excitation signal by low-pass **filtering** (see HF Extraction and Energy Computation).

These speech descriptors are quantized and multiplexed to generate the coded speech data to be **provided** to the speech synthesizer whenever the speech signal needs be reconstructed.

The synthesizer **is** made to perform the following operations:

- decoding and **up**-sampling to 8kHz the Base-Band signal(see BB Decode in Fig.1)

- generating a high frequency signal (1000-3400 Hz) by non-linear distorsion high-pass **filtering** and energy adjustment of the base-band signal (see Non Linear Distortion HP **Filtering** and Energy Adjustment)

- exciting an all-pole prediction **filter** corresponding the vocal tract by the sum of the base-band signal and of the... ...I) and (II) are separately coded. But the third speech descriptors (III) derived through analysis **of** the high and **low frequency** bandwidth contents, differs from the descriptor (III) of a conventional RELP as represented in figure... ...upper-band) (Fig.3d) signals.

The problem faced with RELP vocoders is to derive at **the** receiver end (synthesizer) a synthetic high-frequency signal from the transmitted base-band signal. As... ...making a non-linear distortion of the base-band signal followed by a high-pass **filtering** and a level adjustment according to the transmitted energy. The signal obtained through these operations... pulses of the base-band pulse train.

The upper-band signal y(n) is then **modulated** by the windowing signal w(n-K).

(8) y(")(n) = y(n).w(n-K... ...the upper band according to the pulse/noise model in device (15), as represented in **Fig**.9.

This high-frequency signal s(n) is then added to the delayed base-band signal to obtain the excitation signal of the predictor **filter** to be used for performing the LP Synthesis function of Fig.2.

Fig.9 shows...each pitch period so as to improve the periodicity of the full high-band signal s(n). This reset is **achieved** by the shifted pulse train z(n-K).

The pulse and noise signal components are then summed up and **filtered** by a high-pass **filter 19** which removes the (0-1000Hz) of the upper-band signal s(n). Note on Fig.5 that the delay introduced by the high-pass **filter** on the high-frequency band is compensated by a delay (20) on the base-band...

Claims: ...and a high frequency (HE) bandwidth to be coded separately, said process comprising : - coding said **low frequency** bandwidth signal ; - processing said high frequency bandwidth signal to derive therefrom high frequency

energy information... ...shifting said low frequency bandwidth decoded data using said phase shift information ; - combining said shifted **low frequency** decoded data with **said** high frequency energy data to derive therefrom a synthesized upper band signal ; and - adding said... ...process including : - demultiplexing and decoding said linear parameters ; - using said decoded linear prediction parameters to **adjust** a synthesis **filter** fed with the signal provided by said adding operation.

8. A coding process according to...14, wherein said upper band analysis means include :

- windowing means sensitive to said shifted pulse **train** and to said **pitch** M to derive therefrom a w(n-k) train ;

- modulating means sensitive to said w... ...i) and z(n-K) to derive s(n) ;

- summing means for summing said upper **band** train s(n) and a delayed x(n) train ;

- LP synthesis **filter** tuned by said decoded LP parameters and sensitive to the output of said summing means... ...a noise signal component e'(n) = e(n).E(sup(1/2) ;

- adding means for **adding** said noise component to said pulse signal component ; and,

- high pass **filter** connected to said adding means to provide said s(n).

Claims: ...umfast: Demultiplexen und Decodieren der linearen Parameter, Verwenden der decodierten linearen Pradiktionsparameter, um ein Synthese-**Filter** einzustellen, das mit dem von dem Addierarbeitsgang gelieferten Signal gespeist wird.

8. Verfahren zum Codieren...

13/3K/6 (Item 1 from file: 349) Links
PCT FULLTEXT

01071682
METHOD AND DEVICE FOR EFFICIENT FRAME ERASURE CONCEALMENT IN LINEAR PREDICTIVE BASED SPEECH CODECS
PROCEDE ET DISPOSITIF DE MASQUAGE EFFICACE D'EFFACEMENT DE TRAMES DANS DES CODEC VOCAUX DE TYPE LINEAIRE PREDICTIF

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	Country	Number	Kind	Date
Patent	WO	2003102921	A1	20031211
Application	WO	2003CA830		20030530
Priorities	CA	2388439		20020531

Designated States: (All protection types applied unless otherwise stated - for applications 2004+)

|EP| AT; BE; BG; CH; CY; CZ; DE; DK; EE; ES;
FI; FR; GB; GR; HU; IE; IT; LU; MC; NL;
PT; RO; SE; SI; SK; TR;

|OA| BF; BJ; CF; CG; CI; CM; GA; GN; GQ; GW;
ML; MR; NE; SN; TD; TG;

|AP| GH; GM; KE; LS; MW; MZ; SD; SL; SZ; TZ;
UG; ZM; ZW;

|EA| AM; AZ; BY; KG; KZ; MD; RU; TJ; TM;

Main International Patent Classes (Version 7):

IPC	Level
G10L-019/00	Main

Publication Language: English
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Fulltext word count: 22944

Detailed Description:

...a predetermined number corresponding typically
i 5 to 10-30 ms. A linear prediction (LP) **filter** is computed and transmitted every frame. The computation of the LP **filter** typically needs a lookahead, a 5-15 ms speech segment from the subsequent frame. The... ...the decoder, where the reconstructed excitation signal is used as the input of the LP **filter**.

As the main applications of low bit rate speech encoding are wirel ess
mobile communication... ...telephone network) that uses the legacy narrow band speech signals.

The adaptive codebook, or the **pitch predictor**, in CELP plays an
important role in maintaining high speech quality at low bit rates ... ...be missing from the adaptive codebook content. This will have a severe effect on the **pitch predictor** in consequent good frames, resulting in long time before the synthesis signal converge to the... ...the AMR-WB encoder of Figure
2, wherein, the down-sampler module, the high-pass **filter** module and the preemphasis **filter** module have been grouped in a single pre-processing module, .and wherein the closed-loop... ...optional pre-processing module
202. Pre-processing module 202 may consist of a high-pass **filter** with a 50 Hz cut-off frequency. High-pass **filter** 202 removes the unwanted sound components below 50 Hz.

The down-sampled, pre-processed signal... ...at a sampling frequency of 12.8
kHz). In an illustrative embodiment of the preemphasis **filter** 203, the signal sp(n) is preemphasized using a **filter** having the following transfer function.

P(Z) = I - Yz-1
where p is a preemphasis... ...0 and I (a typical value is p = 0.7). The function of the preemphasis **filter** 203 is to enhance the high frequency contents of the input speech signal. It also... ...quality. This will be explained in more detail herein below.

The.output of the preemphasis **filter** 203 is denoted s(n). This signal is
used for performing LP analysis in module... ...are computed from the windowed signal, and Levinson-Durbin recursion is used to compute LP **filter** coefficients, ai, where i@-- I I... p, and where p is the LP order, which... ...The parameters ai are the coefficients of the transfer function A(z) of the LIP **filter**, which is given by the following relation.

P
A(z) = I + a iz
LP analysis is performed in module 204, which also performs the quantization and interpolation of the LP **filter** coefficients. The LP **filter** coefficients are first transformed into another equivalent domain more suitable for quantization and interpolation purposes... ...are two domains in which quantization and interpolation can be efficiently performed. The 16 LID **filter** coefficients, ab can be quantized in the order of 30 to 50 bits using split... ...or a combination thereof. The purpose of the interpolation is to enable updating the LP **filter** coefficients every subframe while transmitting them once every frame, which improves the encoder performance without increasing the bit rate. Quantization and interpolation of the LP **filter** coefficients is believed to be otherwise well known to those of ordinary skill in the... ...64 samples at the sampling frequency of 12.8 kHz). In the following description, the **filter** A(z) denotes the unquantized interpolated LP **filter** of the subframe, and the **filter** A(z) denotes the quantized interpolated LP'**filter** of the subframe. The **filter** AW is supplied every subframe to a multiplexer 213 for transmission through a... ... weighted domain. The weighted signal sw(n) is computed in a perceptual weighting **filter** 205 in response to the signal s(n) from the pre-emphasis **filter** 203. A perceptual weighting **filter** 205 with fixed denominator, suited for wideband signals, is used. An example of transfer function for the perceptual weighting **filter** 205 is given by the following relation.

W(z) = A(z/y,)/(1 - Y2Z-') where... ...computed. This is usually done by subtracting the zero-input response so of weighted synthesis **filter** W(z)IA(z) from the weighted speech signal sw(n). This zero-input response... ...calculated by a zero-input response calculator 208 in response to the quantized interpolation LP **filter** A(z) from the LP analysis, quantization and interpolation module 204 and to the initial states of the weighted synthesis **filter** W(z)IA(z) stored in memory update module 211 in response to the LP **filters** A(z) and A(z), and the excitation vector u. This operation is well known... ...not be further described.

A N-dimensional impulse response vector h of the weighted synthesis **filter** W(z)IA(z) is computed in the impulse response generator 209 using the coefficients of the LP **filter** A(z) and A(z) from module 204. Again, this operation is well known to... ...finding the best pitch lag T and gain b that minimize a mean squared weighted **pitch prediction** error, for example
e ')=Ilx-b I)y ')II' where j-- I I 21 ... I k
between the target vector x and a scaled **filtered** version of the past excitation.

More specifically, in the present illustrative implementation, the pitch (pitch... ...15), which significantly simplifies the search procedure. A simple procedure is used for updating the **filtered** codevector YT (this vector is defined in the following description) without the need to compute... ...spectrum. This is achieved by processing the pitch codevector through a plurality of frequency shaping **filters** (for example low-pass or band-pass **filters**). And the frequency shaping **filter** that minimizes the mean-squared weighted error eu) is selected. The selected frequency shaping **filter** is identified by an indexj.

The pitch codebook index T is encoded and transmitted to... ...
x'= x - bYT
where b is the pitch gain and YT is the **filtered** pitch codebook vector (the past excitation at delay T **filtered** with the selected frequency shaping **filter** (index J) **filter** and convolved with the impulse response h).

The innovative excitation search procedure in CELP is... ...which minimize the mean-squared error E between the target vector x" and a scaled **filtered** version of the codevector ck, for example.

E=Jjx@-gHck 112
where H is a... ...codebook is a dynamic codebook consisting of an algebraic codebook followed by an adaptive pre-**filter** F(z) which enhances special spectral components in order to improve the synthesis speech quality... ...excitation signal u improves the quality of voiced segments. The periodicity enhancement is achieved by **filtering** the innovative codevector ck from the innovation (fixed) codebook through an innovation **filter** F(z) (pitch enhancer 305) whose frequency response emphasizes the higher frequencies more than the lower frequencies. The coefficients of the innovation **filter** F(z) are related to the amount of periodicity in the excitation signal u.

An efficient, illustrative way to derive the coefficients of the innovation **filter** F(z) is to relate them to the amount of pitch contribution in the total... ...higher frequencies are more strongly emphasized (stronger overall slope) for higher

pitch gains. The innovation **filter** 305 has the effect of lowering the energy of the innovation codevector ck at lower... ...signal u at lower frequencies more than higher frequencies. A suggested form for the innovation **filter** 305 is the following.

FW = -az + I - ciz
where a is a periodicity factor derived... ...to produce a pitch codevector.

The pitch codevector is then processed through a low-pass **filter** 302 whose cutoff frequency is selected in relation to index j from the demultiplexer 317 to produce the **filtered** pitch codevector vT. Then, the **filtered** pitch -codevector vT is then amplified by the pitch gain b by an amplifier 326... ...and 0.25 for purely voiced signals.

The enhanced signal cf is therefore computed by **filtering** the scaled innovative codevector gck through the innovation **filter** 305 (F(z)).

The enhanced excitation signal W is computed by the adder 320 as... ...and the enhanced excitation signal W is used at the input of the LIP synthesis **filter** 306.

The synthesized signal s'is computed by **filtering** the enhanced excitation signal W through the LP synthesis **filter** 306 which has the form 1 I A(z), where A(z) is the quantized, interpolated LP **filter** in the current subframe. As can be seen in Figure 3, the quantized, interpolated LP... ...A(z) on line 325 from the demultiplexer 317 are supplied to the LIP synthesis **filter** 306 to adjust the parameters of the LP synthesis **filter** 306 accordingly. The deemphasis **filter** 307 is the inverse of the preemphasis **filter** 203 of Figure 2. The transfer function of the deemphasis **filter** 307 is given by
D(z) = 1/(1 - I-iz-')
where p is a preemphasis... ...located between 0 and 1 (a typical value is p = 0.7). A higher-order **filter** could also be used.

The vector sy is **filtered** through the deemphasis **filter** D(z) 307 to obtain the vector sd, which is processed through the high-pass **filter** 308 to remove the unwanted frequencies below 50 Hz and further obtain sh.

The oversampler... ...310 and requires input from voicing factor generator 304 (Figure 3).

The resulting band-pass **filtered** noise sequence z from the high frequency generation module 310 is added by the adder... ...Parameter Bits I Frame
LP Parameters 46
Pitch Delay 30= 9 + 6 + 9 + 6
Pitch **Filtering** 4= 1 + I + 1 + 1
Gains 28= 7 + 7 + 7 + 7
Algebraic Codebook 144= 36...desynchronized from the encoder. The main reason is that low bit rate encoders rely on **pitch prediction**, and during erased frames, the memory of the **pitch predictor** is no longer the same as the one at the encoder. The problem is amplified... ...the AMR-WB encoder 400. In this simplified block diagram, the downsampler 201, high-pass **filter** 202 and preemphasis **filter** 203 are grouped together in the preprocessing module 401.

Also, the closed-loop search module... ...In the present illustrative embodiment, the spectral tilt is estimated as a ratio between the **energy** concentrated in **low frequencies** and the **energy** concentrated in high frequencies. However, it can also be estimated in different ways such as... ...bands have been excluded from the computation to improve the discrimination between frames with high

energy concentration in **low frequencies** (generally voiced) and with high energy concentration in high frequencies (generally unvoiced). In between, the ... ...for any of the classes and would increase the decision confusion.

In module 500, the **energy** in **low frequencies** is computed differently for long pitch periods and short pitch periods. For voiced female speech... ...to the nearest harmonics are taken into account. Hence, if the structure is harmonic in **low frequencies**, only high **energy** term will be included in the sum. On the other hand, if the structure is... ...will be random and the sum will be smaller. Thus even unvoiced sounds with high **energy** content in **low frequencies** can be detected. This processing cannot be done for longer pitch periods, as the frequency... ...and also for a priori unvoiced sounds (i.e.

when rx+re<0.6), the **low frequency energy** estimation is done per critical band and is computed as

9
El e(i)
10... ...of the weighted speech signal sw(n) of the current frame from the perceptual weighting **filter** 205 and Ee is the energy of the error between this weighted speech signal and the weighted synthesis signal of the current frame from the perceptual weighting **filter** 205'.

I 0
The pitch stability counter PC assesses the variation of the pitch period... ...be
used at the decoder to help the classification, as the parameters of the LP **filter** or the pitch stability.

In case of source-controlled variable bit rate coder, the information... ...if the energy control is most important for voiced speech because of the long term **prediction** (**pitch prediction**), it is important also for unvoiced speech. The reason here is the prediction of the... ...domain has the disadvantage of not taking into account the influence of the LP synthesis **filter**. This can be particularly tricky in the case of voiced recovery after several lost voiced... ...is typically used during the concealment with some attenuation strategy. When a new LIP synthesis **filter** arrives with the first good frame after the erasure, there can be a mismatch between the excitation energy and the gain of the LIP synthesis **filter**. The new synthesis **filter** can produce a synthesis signal with an energy highly different from the energy of the...obtained when the position of the first glottal pulse is measured on the low-pass **filtered** residual signal.

The position of the first glottal pulse is coded using 6 bits in... ...be however easily applied to
any speech codec where the synthesis signal is generated by **filtering** an
excitation signal through an LP synthesis **filter**. The concealment strategy can be summarized as a convergence of the signal energy and the... ...attenuation factor a. The factor a is further dependent on the stability of the LP **filter** for UNVOICED frames. In general, the convergence is slow if the last good received frame... ...A stability factor 0 is computed based on a distance measure between the
adjacent LP **filters**. Here, the factor 0 is related to the ISF (Immittance Spectral Frequencies) distance measure and... ...1 st erased frame after a good frame, this pitch pulse is first low-pass **filtered**. The **filter** used is a 5 simple 3-tap linear phase FIR **filter** with **filter** coefficients equal to- 0. 1 8, 0.64 and
If a voicing information is available, the **filter** can be also selected
dynamically with a cut-off frequency dependent on the voicing.

The... ...correctly received or non erased) received frame is different from UNVOICED, the innovation excitation is **filtered** through a linear phase FIR high-pass **filter** with coefficients 01-25, 109, 0.7813, 109,
To decrease the amount of noisy components during voiced segments,
these **filter** coefficients are multiplied by an adaptive factor equal to (0.75 - 0.25 rv)7... ...is available.

Spectral Envelope Concealment, Synthesis and updates
To synthesize the decoded speech, the LIP **filter** parameters must be
obtained. The spectral envelope is gradually moved to the estimated envelo pe... ...ISF of the estimated comfort noise envelope and p is the order of the LP **filter**.

The synthesized speech is obtained by **filtering** the excitation signal
through the LIP synthesis **filter**. The **filter** coefficients are computed from the ISF representation and are interpolated for each subfrarne (four (4... ...are using the past excitation signal to encode the present frame excitation (long-term or **pitch prediction**). Also, most of the quantizers (ILP quantizers, gain quantizers) make use of a prediction.

Artificial... ...lost onset, the periodic part of the excitation is constructed artificially as, a low-pass **filtered** periodic train of pulses separated by a pitch period. In the present illustrative embodiment, the low-pass **filter** is a simple linear phase FIR **filter** with the impulse response hlow = @ 01 25, 0.109, 0.7813, 0.109, 01251. However,
the **filter** could be also selected dynamically with a cut-off frequency
corresponding to the voicing information... ...pitch periods of all subframes where the artificial onset reconstruction is used.

The low-pass **filtered** impulse train is realized by placing the impulse responses of the low-pass **filter** in the adaptive excitation buffer (previously initialized to zero), The first impulse response will be... ...defined in Equations 16 and 17) and divided by the gain of the LIP synthesis **filter**. The LP synthesis **filter** gain is computed as.

9LP h 2 (i)
i=O (31)
where h(i) is the LP synthesis **filter** impulse response. Finally, the artificial onset gain is reduced by multiplying the periodic part with... ...of the artificial onset and the regular CELP decoding could be used instead.

The LIP **filter** for the output speech synthesis is not interpolated in the case of an artificial onset... ...frame is typically used during the concealment with some attenuation strategy. When a new LIP **filter** arrives with the first good frame after the erasure, there can be a mismatch between the excitation energy and the gain of the new LIP synthesis **filter**. The new synthesis **filter** can produce a synthesis signal with an energy highly different from the energy of the... ...implementation.

Conducting frame erasure concealment and decoder recovery comprises,
when a gain of a LIP **filter** of a first non erased frame received following frame erasure is higher than a gain of a LP **filter** of a last frame erased during said frame erasure, adjusting the energy of an LP **filter** excitation signal produced in the decoder during the received first non erased frame to a gain of the LIP **filter** of said received first non erased frame using the following relation.

If Eq cannot be... ...be taken because
of the possible mismatch,between the excitation signal energy and the LIP **filter** gain, mentioned previously. A particularly dangerous situation arises when the gain of the LIP **filter** of a first non erased frame received following frame erasure is higher than the gain of the LP **filter** of a last frame erased during that frame erasure. In that particular case,,the energy of the LP **filter** excitation signal produced in the decoder during the received first non erased frame is adjusted to a gain of the LIP **filter** of the received first non erased frame using the following relation.

Eq =El ELpo
ELPI
where ELpO is the energy of the LP **filter** impulse response of the last good frame before the erasure and ELp1 is the energy of the LIP **filter** of the first good frame after the erasure. In this implementation, the LIP **filters** of the last subframes in a frame are used. Finally, the value of Eq is...

Claims:

...defined in claim 13, comprising estimating the spectral tilt parameter as a ratio between an **energy** concentrated in **low frequencies** and an **energy** concentrated in high frequencies.

17 A method as defined in claim 13, comprising estimating the...a non erased unvoiced frame after frame erasure, generating no periodic part of a LP **filter** excitation signal;following receiving, after frame erasure, of a non erased frame other than unvoiced, constructing a periodic part of the LP **filter** excitation signal by repeating a last pitch period of a previous frame.
27 A method as defined in claim 26, wherein constructing the periodic
part of the LP **filter** excitation signal comprises **filtering** the repeated last pitch period of the previous frame through a low-pass **filter**.
28 A method as defined in claim 27, wherein:
determining concealment/recovery parameters comprises computing avoicing information parameter;the low-pass **filter** has a cut-off frequency; andconstructing the periodic part of the excitation signal comprises... ...and decoder recovery comprises randomly generating a non1 5 periodic, innovation part of a LIP **filter** excitation signal.
30 A method as defined in claim 29, wherein randomly generating the
non-periodic, innovation part of the LP **filter** excitation signal comprises generating a random noise.
31 A method as defined in claim 29, wherein randomly generating the
non-periodic, innovation part of the LP **filter** excitation signal comprises randomly generating vector indexes of an innovation codebook.
32 A method as... ...transition, voiced, or onset; and
randomly generating the non-periodic, innovation part of the LIP **filter**excitation signal further comprises:if the last correctly received frame is different from unvoiced,**filtering** the innovation part of the excitation signal through a high pass**filter**; andif the last correctly received frame is unvoiced, using only theinnovation part of... ...lost onset by constructing a periodic part of an excitation signal as a low-pass **filtered** periodic train of pulses separated by a pitch period.
34 A method as defined in... ...and
conducting frame erasure concealment and decoder recovery comprises,when a gain of a LP **filter** of a first non erased frame received following frame erasure is higher than'a gain of a LP **filter** of a last frame erased during said frame erasure, adjusting the energy of an LP **filter** excitation signal produced in the decoder during the received first non erased frame to a gain of the LP

filter of said received first non erased frame.
40 A method as defined in claim 39 wherein:
adjust ing the energy of an LP **filter** excitation signal produced in thedecoder during the received first non erased frame to a gain of the LP **filter** of said received first non erased frame comprises using the following relation:.1=1 ELPO...

...of the current frame, ELpo is the energy of an impulse response of the LIP **filter** to the last non erased frame received before the frame erasure, and ELpl is the energy of the impulse response of the LIP **filter** to I 0 the received first non erased frame following frame erasure.
41 A method... ...a 'non erased unvoiced frame after frame erasure,
generating no periodic part of a LP **filter** excitation signal;following receiving, after frame erasure, of a non erased frame other than unvoiced, constructing a periodic part of the LP **filter** excitation signal by repeating a last pitch period of a previous frame.
48 A method as defined in claim 47, wherein constructing the periodic
part of the excitation signal comprises **filtering** the repeated last pitch period of the previous frame through a low-pass **filter**.
49 A method as defined in claim 48, wherein:
determining, in the decoder, concealment/recovery parameters comprisescomputing a voicing information parameter;the low-pass **filter** has a cut-off frequency; andconstructing the periodic part of the LIP **filter** excitation signal comprises dynamically adjusting the cut-off frequency in relation to the voicing information...
...erasureconcealment and decoder recovery comprises randomly generating a nonperiodic, innovation part of a LIP **filter** excitation signal.
51 A method as defined in claim 50, wherein randomly generating the
non-periodic, innovation part of the LIP **filter** excitation signal comprises generating a random noise.
52 A method as defined in claim 50, wherein randomly generating the
non-periodic, innovation part of the LP **filter** excitation signal comprises randomly ,generating vector indexes of an innovation codebook.
53 A method as... ...transition, voiced, or onset; and
randomly generating the non-periodic, innovation part of the LIP **filterexcitation** signal further comprises:if the last received non erased frame is different from unvoiced,**filtering** the innovation part of the LP **filter** excitation signal through a5 high pass **filter**; andif the last received non erased frame is unvoiced, using only theinnovation part of the LP **filter** excitation signal.
54 A method as defined in claim 50, wherein:
the sound signal is... ...lost onset by constructing a periodic part of an excitation signal as a low-pass **filtered** periodic train of pulses separated by a pitch period.
55 A method as defined in... ...frame erasure
concealment and decoder recovery further comprises constructing an innovation part of the LP **filter** excitation signal by means of normal decoding.
56 A method as defined in claim 55, wherein constructing an innovation
part of the LP **filter** excitation signal comprises randomly choosing entries of an innovation codebook.
57 A method as defined... ...and
conducting frame erasure concealment and decoder recovery comprises,when a gain of a LIP **filter** of a first non erased frame received following frame erasure is higher than a gain of a LP **filter** of a last frame erased during said frame erasure, adjusting the energy of an LIP **filter** excitation signal produced in the decoder during the received first non erased frame to a gain of the LIP **filter** ofsaid received first non erased frame using the following relation:Eq =El ELpOELPI... ...of the current frame, ELpO is the energy of an impulse response of the LIP **filter** to the last non erased frame received before the frame erasure, and ELpl is the energy of the impulse response of the LP **filter** to the received first non erased frame following frame erasure.
60 A device for improving... ...claim 72, comprising means for estimating the
spectral tilt parameter as a ratio between - an **energy** concentrated in **low frequencies** and an **energy** concentrated in high frequencies.
76 A device as defined in claim 72, comprising means for ... ...erased unvoiced frame after frame erasure,
means for generating no periodic part of a LP **filter** excitation signal;following receiving, after frame erasure, of a non erased frame other than unvoiced, means for constructing a periodic part of the LP **filter** excitation signal by repeating a last pitch period of a previous frame.I0 86... ...defined in claim 85, wherein the means for constructing the periodic part of the LP **filter** excitation signal comprises a low-pass **filter** for **filtering** the repeated last pitch period of the previous frame.
87 A device as defined in... ...determining concealment/recovery parameters comprises
means for computing a voicing information parameter;the low-pass **filter** has a cut-off frequency; andthe means for constructing the periodic part of the... ...decoder recovery comprises means for randomly generating a non-periodic, innovation part of a LP **filter** excitation signal.
89 A device as defined in claim 88, wherein the means for randomly
generating the non-periodic, innovation part of the LP **filter** excitation signal comprises means for generating a random noise.
90 A device as defined in... ...88, wherein the means for randomly
generating the non-periodic, innovation part of the LP **filter** excitation signalcomprises means for randomly generating vector indexes of an innovationcodebook.
91 A... ...onset; and

the means for randomly generating the non-periodic, innovation part of theLP **filter** excitation signal further comprises:if the last correctly received frame is different from unvoiced, ahigh-pass **filter** for **filtering** the innovation part of the excitation signal;andif the last correctly received frame is... ...for conducting frame erasure concealment and decoderrecovery comprises, when a gain of a LIP **filter** of a first non erased frame received following frame erasure is higher than a gain of a LP **filter** of a last frame erased during said frame erasure, means for adjusting the energy of an LIP **filter** excitation signal produced in the decoder during the received first non erased frame to a gain of the LP **filter** of said received first non erased frame.

99 A device as defined in claim 98, wherein:

the means for adjusting the energy of an LIP **filter** excitation signalproduced in the decoder during the received first non erased frame to a gain of 5 the LP **filter** of said received first non erased frame comprises means for usingthe following relation:Eq... ...of the current frame, ELpO is the energy of an impulse response of the LP **filter** to the last non erased frame received before the frame erasure, and ELpl is the energy of the impulse response of the LIP **filter** to the received first non erased frame following frame erasure.1 00. A device as...erased unvoiced frame after frame erasure,means for generating no periodic part of a LP **filter** excitation signal;following receiving, after frame erasure, of a non erased frame other than unvoiced, means for constructing a periodic part of the LP **filter** excitation signal by repeating a last pitch period of a previous frame. 107. A device... ...the means for constructing the periodic part of the excitation signal comprises a low-pass **filter** for **filtering** the repeated last pitch period of the previous frame.108. A device as defined in... ...decoder, concealment/recoveryparameters comprises means for computing a voicing information parameter;the low-pass **filter** has a cut-off frequency;.andthe means for constructing the periodic part of the LIP **filter** excitation signal comprises means for dynamically adjusting the cut-off frequency in relation to the... ...decoder recovery comprises means for randomly generating a non-periodic, innovation part of a LIP **filter** excitation signal.1 1 0. A device as defined in claim 109, wherein the means for randomlygenerating the non-periodic, innovation part of the LP **filter** excitation signal 1 5 comprises means for generating a random noise.1 1 1. A... ...109, wherein the means for randomlygenerating the non-periodic, innovation- part of the LIP **filter** excitation signalcomprises means for randomly generating vector indexes of an innovationcodebook.112. A... ...onset;andthe means for randomly generating the non-periodic, innovation part of theLIP **filter** excitation signal further comprises:if the last received non erased frame is different from unvoiced,a high-pass **filter** for **filtering** the innovation part of the LP **filter**excitation signal; andif the last received non erased frame is unvoiced, means forusing only the innovation part of the LP **filter** excitation signal.1 1 3. A device as defined in claim 109, wherein:the sound... ...lost onset by constructing a periodic part of an excitation signal as a low-pass **filtered** periodic train of pulses separated by a pitch period.114. A device as defined in... ...concealment and decoder recovery further comprises means for constructing an innovation part of the LP **filter** excitation signal by means of normal decoding. 115. A device as defined in claim 114, wherein the means for constructing an innovation part of the LP **filter** excitation signal comprises means for randomly choosing entries of an innovation codebook. 116. A device... ...for conducting frame erasure concealment and decoderrecovery comprises, when a gain of a LP **filter** of a first non erased frame received following frame erasure is higher than a gain of a LP **filter** of a last frame erased during said frame erasure, means for adjusting the energy of an LIP **filter** excitation signal produced in the decoder during the received first non erased frame to a gain of the LP **filter** of said received first non erased frame using thefollowing relation:Eq =El ELpOELP1... ...of the current frame, ELpO is the energy of an impulse response of the LP **filter** to.the last non erased frame received before the frame erasure, and ELpl is the energy of the impulse response of the LP **filter**

to the received first non erased frame following frame erasure.11 9. A system for...

13/3K/7 (Item 2 from file: 349) **Links**

PCT FULLTEXT

00561930

PERIODICITY ENHANCEMENT IN DECODING WIDEBAND SIGNALS

AMELIORATION DE LA PERIODICITE DANS LE DECODAGE DE SIGNAUX A LARGE BANDE

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English Abstract:

...in relation to a pitch codevector and an innovative codevector for supplying a signal synthesis **filter** in view of producing a synthesized wideband signal. In this periodicity enhancing device and method... ...is responsive to the adaptive and innovative codevectors for calculating a periodicity factor. An innovation **filter** subsequently processes the innovative codevector in relation to this periodicity factor to reduce **energy** of a **low frequency** portion of the innovative codevector and enhance periodicity of a low frequency portion of the excitation signal. As an example, the innovation **filter** has a transfer function of the form: i(F)(z)=-alpha(i(z))+1-alpha...

Detailed Description:

...to a method and device for
enhancing periodicity of the excitation of a signal synthesis **filter** in view of producing a synthesized wideband signal.

2. Brief description of the prior art... ...and wireless applications, as well as Internet and packet network applications. Until recently, telephone bandwidths **filtered** in the range 200-3400 Hz were mainly used in speech coding applications. However, there... ...number (corresponding to 10-30 ms of speech). In CELP, a linear prediction (LP) synthesis **filter** is computed and transmitted every frame. The L-sample frame is then divided into smaller... ...signal is transmitted and used at the decoder as the input of the LP synthesis **filter** in order to obtain the synthesized speech.

An innovative codebook in the CELP context, is... ...synthesize speech according to the CELP technique, each block of N samples is synthesized by **filtering** an appropriate codevector from a codebook through time varying **filters** modeling the spectral characteristics of the speech signal. At the encoder end, the synthesis output... ...perceptually weighted distortion
measure. This perceptual weighting is performed using a so-called
perceptual weighting **filter**, which is usually derived from the LIP synthesis 1 5 **filter**.

The CELP model has been very successful in encoding
telephone band sound signals, and several... ...improves the
quality in case of voiced segments. This was done in the past by **filtering** the
innovative codevector from the fixed codebook through a **filter** having a
transfer function of the form 1 /(1 -ebz-T) where e is a... ...invention is to propose a new alternative
approach by which periodicity enhancement is achieved through **filtering** the innovative codevector by an innovation **filter** which reduces the low1 5 frequency contents of the innovative codevector, whereby the innovative contribution...
...in relation to a pitch codevector and an innovative codevector for supplying a signal synthesis **filter** in view synthesizing a wideband signal. In
this periodicity enhancing method, a periodicity factor related to the
wideband signal is calculated. Then, the innovative codevector is **filtered** in relation to the periodicity factor to thereby reduce **energy** of a **low frequency**

portion of the innovative codevector and enhance periodicity of a low frequency portion of the... ...excitation signal produced in relation to adaptive and innovative codevectors for supplying a signal synthesis **filter** in view of synthesizing a wideband signal, comprises.

a) a factor generator for calculating a periodicity factor related to 0 said wideband signal; and
b) an innovative **filter** for **filtering** the innovative codevector in relation to the periodicity factor to thereby reduce **energy** of a **low frequency** portion of the innovative codevector and enhance periodicity of a low frequency portion of the excitation signal.

5
According to a first preferred embodiment.

- the innovative codevector is **filtered** with a transfer function of the form.

F(z) = -az + 1 -az
where a is... ...of the innovative codevector.

According to a second preferred embodiment.

- the the innovative codevector is **filtered** with a transfer function of the form.

F(z) =1 -Oz -1
where a is... ...from this encoded wideband signal at least pitch codebook parameters, innovative codebook parameters, and synthesis **filter** coefficients;
b) an pitch codebook responsive to the pitch codebook parameters for producing a pitch... ...factor generator for calculating a periodicity factor related to the wideband signal; and the innovation **filter** for **filtering** the innovative codevector in relation to the periodicity factor;
e) a combiner circuit for combining the pitch codevector and the innovative codevector **filtered** by the innovation **filter** to thereby produce a 0 periodicity-enhanced excitation signal; and
f) a signal synthesis **filter** for **filtering** that periodicity-enhanced excitation signal in relation to the synthesis **filter** coefficients to thereby produce the synthesized wideband signal.

5 According to the present invention, in... ...from this encoded wideband signal at least pitch codebook parameters, innovative codebook parameters, and synthesis **filter** coefficients; an pitch codebook responsive to the pitch codebook parameters for producing a pitch codevector... ...codevector and the innovative codevector to thereby produce an
excitation signal; and a signal synthesis **filter** for **filtering** that excitation signal in relation to the synthesis **filter** coefficients to thereby produce the synthesized wideband signal;
the improvement therein comprising a periodicity enhancing... ...factor generator for calculating a periodicity factor related to the wideband signal; and the innovation **filter** for **filtering** the innovative codevector in relation to the periodicity factor before supplying this innovative codevector to...and below such as Code-Excited Linear Prediction (CELP)
encoders typically use a LP synthesis **filter** to model the short-term spectral 1 5 envelope of the voice signal. The LP... ...signal in the frame are computed, encoded, and
transmitted. LP parameters representing the LP synthesis **filter** are usually computed once every frame. The frame is further divided into smaller blocks of... ...pre
processing, and preemphasis);
s, Weighted speech vector;
so Zero-input response of weighted synthesis **filter**;
sp Down-sampled pre-processed signal;
Oversampled synthesized speech signal;
s' Synthesis signal before deemphasis... ...x Target vector for pitch search;
X' Target vector for innovation search;
h Weighted synthesis **filter** impulse response;
vT Adaptive (pitch) codebook vector at delay T;

yT **Filtered** pitch codebook vector (VTconvolved with h);
ck Innovative codevector at index k (k-th entry...codebook index);
b Pitch gain (or pitch codebook gain);
i Index of the low-pass **filter** used on the pitch codevector;
k Codevector index (innovation codebook entry); and
9 Innovation codebook... ...optional pre-processing block
102. Pre-processing block 102 may consist of a high-pass **filter** with a 50 Hz cut-off frequency. High-pass **filter** 102 removes the unwanted sound components below 50 Hz.

The down-sampled pre-processed signal... ...at a sampling frequency of 12.8 kHz). In a preferred embodiment of the preemphasis **filter** 103, the signal sp(n) is preemphasized using a **filter** having the following transfer function.

P(Z) PZ

where u is a preemphasis... ...value located between 0 and 1 (a typical value isu = 0.7). A higher-order **filter** could also be used. It should be pointed out that high-pass **filter** 102 and preemphasis **filter** 103 can be interchanged to obtain more efficient fixed-point implementations.

The function of the preemphasis **filter** 103 is to enhance the high frequency contents of the input signal. It also reduces... ...quality. This will be explained in more detail herein below.

The output of the preemphasis **filter** 103 is denoted s(n). This signal is used for performing LIP analysis in calculator... ...are computed from the windowed signal, and Levinson-Durbin recursion is used to compute LP **filter** coefficients, aj, where and where p is the LIP order, which is typically 16 in wideband coding. The parameters ai are the coefficients of the transfer function of the LIP **filter**, which is given by the following relation.

P
A(z) = 1 +Ya Z
LIP analysis... ...performed in calculator module 104, which also performs the quantization and interpolation of the LP **filter** coefficients. The LIP **filter** coefficients are first transformed into another equivalent domain more suitable for quantization and interpolation purposes... ...are two domains in which quantization and interpolation can be efficiently performed. The 16 LIP **filter** coefficients, aj, can be quantized in the order of 30 to 50 bits using split... ...or a combination thereof. The purpose of the interpolation is to enable updating the LIP **filter** coefficients every subframe while transmitting them once every frame, which improves the encoder performance without increasing the bit rate. Quantization and interpolation of the LIP **filter** coefficients is believed to be otherwise well known to those of ordinary skill in the... ...rest of the coding operations performed on a subframe basis. In the following description, the **filter** A(z) denotes the unquantized interpolated LP **filter** of the subframe, and the **filter** A(z) denotes the quantized interpolated LP **filter** of the subframe.

Perceptual Weighting.

In analysis-by-synthesis encoders, the optimum pitch and innovation... ...and weighted synthesis speech.

The weighted signal sjn) is computed in a perceptual weighting **filter** 105. Traditionally, the weighted signal sjn) is computed by a weighting **filter** having a transfer function IM(z) in the form.

W(Z)=A(Zlyd / A(Z of the perceptual weighting **filter** 105. This result is well described by B.S. Atal and M.R. Schroeder in... ...of weighting is controlled by the factors y, and Y2
The above traditional perceptual weighting **filter** 105 works well with telephone band signals. However, it was found that this traditional perceptual weighting **filter** 105 is not suitable for efficient perceptual weighting of wideband signals. It was also found that the traditional

perceptual weighting **filter** 105 has inherent limitations in modelling the formant structure and the required spectral tilt concurrently... ...range
between low and high frequencies. The prior art has suggested to add a tilt **filter** into W(z) in order to control the tilt and formant weighting of the wideband... ...solution to this problem is, in accordance with the present
invention, to introduce the preemphasis **filter** 103 at the input, compute the LP **filter** A(z) based on the preemphasized speech s(n), and use a modified **filter** W(z) by fixing its denominator.

LP analysis is performed in module 104 on the preemphasized signal
s(n) to obtain the LP **filter** A(z). Also, a new perceptual weighting **filter** 105
with fixed denominator is used. An example of transfer function for the
perceptual weighting **filter** 104 is given by the following relation.

W(Z) = '4 (Zlyd / (1 -Y2Z where O... ...z) is computed based on the preemphasized
speech signal s(n), the tilt of the **filter** IIA(zl*yl) is less pronounced compared
to the case when A(z) is computed based on the original speech. Since
deemphasis is performed at the decoder end using a **filter** having the transfer function.

1 5
P =1 1 -fLz
the quantization error spectrum is shaped by a **filter** having a transfer
function Wl(z)P-l(z). When Y2'S set equal to... ...which is typically the case, the spectrum of the quantization error is shaped by a **filter** whose transfer
function is 1 1A(zlyl), with A(z) computed based on the preemphasized
speech... ...this structure for achieving
the error shaping by a combination of preemphasis and modified weighting
filtering is very efficient for encoding wideband signals, in addition to the advantages of ease of... ...computed. This is usually done by subtracting the zero-input response so of weighted synthesis **filter** W(z)IA(z) from the weighted speech signal s"' (n).

This zero-input response... ...the weighted speech
vector in the subframe, and so is the zero-input response of **filter** W(z)IA(z) which is the output of the combined **filter** wwlA(z) due to its initial states.

The zero-input response calculator 108 is responsive to the quantized
interpolated LP **filter** A(z) from the LIP analysis, quantization and interpolation calculator 104 and to the initial states of the weighted synthesis **filter** W(z)IA(z) stored in memory module 1`1 1 to calculate the zero... ...due to the initial states as determined by setting the inputs equal to zero) of **filter** W(z)IA(z). This operation is well known to those of ordinary skill in... ...the target vector x.

A N-dimensional impulse response vector h of the weighted
synthesis **filter** w(z)IA(z) is computed in the impulse response generator 109 5 using the LIP **filter** coefficients A(z) and A(z) from module 104. Again, this operation is well known... ...the impulse response vector h and the open-loop pitch lag ToLas inputs. Traditionally, the **pitch prediction** has been represented by a pitch **filter** having the following transfer function.

1 / (1 -bZ -T)
where b is the pitch gain...a new sample). For pitch lags
T>N, the pitch codebook is equivalent to the **filter** structure (1 1(1-bz-T), and
an pitch codebook vector vAn) at pitch lag T... ...from the past excitation until the vector is completed (this is not equivalent to the **filter** structure).

In recent encoders, a higher pitch resolution is used which
significantly improves the quality... ...voiced sound segments. This is
achieved by oversampling the past excitation signal using polyphase
interpolation **filters**. In this case, the vector v4n) usually corresponds to an interpolated version of the past...
...minimize the mean squared weighted error E between the target vector x and the scaled **filtered** past excitation. Error E being expressed as.

0
E=jjX-bYT 1 12
where yTis the **filtered** pitch codebook vector at pitch lag T-.

n
YT (n) = VT (n) * h(n... ...5), which significantly simplifies the search procedure. A simple procedure is used for updating the **filtered** codevector yTwithout the need to compute the convolution for every pitch lag.

Once an optimum... ...the search (module 107) tests the fractions around that optimum integer pitch lag.

When the **pitch predictor** is represented by a **filter** of the form
1/(1 -bz-T), which is a valid assumption for pitch lags T>N, the spectrum of the pitch **filter** exhibits a harmonic structure over the entire frequency range, with
a harmonic frequency related to... ...to achieve efficient representation of the pitch contribution in voiced segments of wideband speech, the **pitch prediction filter** needs to have the flexibility of varying the amount of periodicity over the wideband spectrum... ...of wideband signals is disclosed in the present specification, whereby several forms of low pass **filters** are applied to the past excitation and the low pass **filter** with higher prediction gain is selected.

When subsample pitch resolution is used, the low pass **filters** can be incorporated into the interpolation **filters** used to obtain the higher pitch resolution. In this case, the third stage of the... ...fractions around the chosen integer pitch lag are tested, is repeated for the several interpolation **filters** having different low-pass characteristics and the fraction and **filter** index which maximize the search criterion C are selected.

A simpler approach is to complete ... ...three stages described above to determine the optimum fractional pitch lag using only one interpolation **filter** with a certain frequency response, and select the optimum low-pass **filter** shape at the end by applying the different predetermined lowpass **filters** to the chosen pitch codebook vector vT and select the low-pass **filter** which minimizes the **pitch prediction** error. This approach is discussed in detail below.

Figure 3 illustrates a schematic block diagram... ...vector vT corresponds to the interpolated past excitation signal. In this preferred embodiment, the interpolation **filter** (in module 301, but not shown) has a low-pass **filter** characteristic removing the frequency contents above 7000 Hz.

In a preferred embodiment, K **filter** characteristics are used; these **filter** characteristics could be low-pass or band-pass **filter** characteristics.

Once the optimum codevector VT is determined and supplied by the pitch codevector generator 302, K **filtered** versions of VT are computed respectively using K different frequency shaping **filters** such as 3050), where j--I, 2, ..., K. These **filtered** versions are denoted VfP , wherej=1, 2, ..., K.

The different vectors VP are convolved in... ...response h to obtain the vectors y6), where
j=01 11 Z I K. To **calculate** the mean squared **pitch prediction** error for each vector /), the value y6) is multiplied by the gain b by means ... ...vector x by means of a corresponding subtractor 3080). Selector 309 selects the frequency shaping **filter** 3050) which minimizes the mean squared **pitch prediction** error
e O = Jjx -b (I)y (j) 11 2 j--1 I 21...,K
To **calculate** the mean squared **pitch prediction** error eO) for each value of yO, the value yO) is multiplied by the gain... ...bG) is calculated in a corresponging gain calculator 3060) in association with the frequency shaping **filter** at index j, using the following relationship.

b U) =x lyUl ljjyU)jj2
In... ...T, and j are chosen based on vT or vfO) which minimizes the mean squared **pitch prediction** error e.

Referring back to Figure 1, the pitch codebook index T is encoded and... ...approach, extra information is needed to encode the index j of the selected frequency shaping **filter** in multiplexer 1 1 2. For example, if three **filters** are used Y=O, 11 21 3) , then two bits are needed to represent this

information. The **filter** index information j can also be encoded jointly with the pitch gain b.

Innovative codebook... ...0
x'=x -b
YT
where b is the pitch gain and yT is the **filtered** pitch codebook vector (the past excitation at delay T **filtered** with the selected low pass **filter** and convolved with the inpulse response h as described with reference to Figure 3).

The... ...gain g which minimize the mean-squared error
between the target vector and the scaled **filtered** codevector
E = 11 x'- gHck 112
where H is a lower triangular convolution matrix derived... ...update.

In memory module 1 1 1 (Figure 1), the states of the weighted
synthesis **filter** W(z)/A(z) are updated by **filtering** the excitation signal u = gck+ bvTthrough the weighted synthesis **filter**. After this **filtering**, the states of the **filter** are memorized and used in the next subframe as initial states for computing the zero... ...known to those of ordinary skill in the art can be used to update the **filter** states.

DECODER SIDE
The speech decoding device 200 of Figure 2 illustrates the various
steps... ...scaled codevector 9Ck at the output of the amplifier
224 is processed through a innovation **filter** 205.

Periodicity enhancement.

The generated scaled codevector at the output of the amplifier... ...improves the
quality in case of voiced segments. This was done in the past by **filtering** the innovation vector from the innovative codebook (fixed codebook) 218 through a **filter** in the form 11(1-ebrl) where e is ...which is part of the
present invention, is disclosed whereby periodicity enhancement is
achieved by **filtering** the innovative codevector Ck from the innovative
(fixed) codebook through an innovation **filter** 205 (F(z)) whose frequency
response emphasizes the higher frequencies more than lower
frequencies. The... ...is less than 0.5, then periodicity is low.

Another efficient way to derive the **filter** F(z) coefficients used in a
preferred embodiment, is to relate them to the amount... ...higher
frequencies are more strongly emphasized (stronger overall slope) for
higher pitch gains. Innovation **filter** 205 has the effect of lowering the
energy of the innovative codevector Ck at low......excitation signal u at lower frequencies more than higher frequencies.

Suggested forms for innovation **filter** 205 are
(1) F(z)=1-Cjz-', or (2) F(z)=-o:z+1-o... ...pitch codevector VT from the pitch codebook

201 is then processed through a low-pass **filter** 202 whose cut-off
frequency is adjusted by means of the indexj from the demultiplexer... ...as follows.

(3 = 0.25 (1 + rj.

The enhanced signal cf is therefore computed by **filtering** the
scaled innovative codevector gckthrough the innovation **filter** 205 (F(z)).

The enhanced excitation signal Wis computed by the adder 220
as.

US... ...and the
enhanced excitation signal u' is used at the input of the LP synthesis **filter** 206.

Synthesis and deemphasis
The synthesized signal s' is computed by **filtering** the enhanced
excitation signal Wthrough the LP synthesis **filter** 206 which has the form 11A(z), where A(z) is the interpolated LP **filter** in the current subfrarne. As can be seen in Figure 2, the quantized LP coefficients A(z) on line 225 from

demultiplexer 217 are supplied to the LP synthesis **filter** 206 to adjust the parameters of the LP synthesis **filter** 206 accordingly. The deemphasis **filter** 207 is the inverse of the preemphasis **filter** 103 of Figure 1. The transfer function of the deemphasis **filter** 207 is given by
D (z) = 1 / (1 -pz
where y is a preemphasis factor ...located between 0 and 1 (a typical value is y = 0.7). A higher-order **filter** could also be used.

The vector s'is **filtered** through the deemphasis **filter** D(z) (module 207) to obtain the vector s., which is passed through the high-pass **filter** 208 to remove the unwanted frequencies below 50 Hz and further obtain sh.

Oversampling and... ...then converted to the speech domain, preferably by shaping it with the same LIP synthesis **filter** used for synthesizing the downsampled signal (section).

The high frequency generation procedure in accordance with... ...using the spectral shaper 215. In the preferred embodiment, this is achieved by **filtering** the noise wg through a bandwidth expanded version of the same LIP synthesis **filter** used in the down-sampled domain (1 1A(zlO.8)).

The corresponding bandwidth expanded LIP **filter** coefficients are calculated in spectral shaper 215.

The **filtered** scaled noise sequence wf is then band-pass **filtered** to the required frequency range to be restored using the band-pass **filter** 216. In the preferred embodiment, the band-pass **filter** 216 restricts the noise sequence to the frequency range 5 7.2 kHz. The resulting band-pass **filtered** noise sequence z is added in adder 221 to the oversampled synthesized speech signal (section...

Claims:

...in
relation to a pitch codevector and an innovative codevector for supplying asignal synthesis **filter** in view of synthesizing a wideband signal, saidperiodicity enhancing device comprising:a) a factor generator for calculating a periodicity factor related tothe wideband signal; andb) an innovation **filter** for **filtering** the innovative codevector inrelation to said periodicity factor to thereby reduce **energy** of a **low frequency** portion of the innovative codevector and enhance periodicity of a low frequency portion... ...innovative codevector.
3 A periodicity enhancing device as defined in claim 1 , wherein said innovation **filter** has a transfer function of the form:F(z) = -az + 1 -az -1where a... ...innovative codevector.
7 A periodicity enhancing device as defined in claim 1 , wherein said innovation **filter** has a transfer function of the form:F(z) = 1 -(Jzwhere a is a... ...to a pitch codevector and an innovative codevector for supplying a signal synthesis **filter** in view of synthesizing a wideband signal, saidperiodicity enhancing method comprising:a) calculating a periodicity factor related to the wideband signal;andb) **filtering** the innovative codevector in relation to said periodicityfactor to thereby reduce **energy** of a **low frequency** portion of the innovativecodevector and enhance periodicity of a low frequency portion of the... ...codevector.
13 A method for enhancing periodicity as defined in claim 1 0, wherein said **filtering** comprises processing the innovation vector through an innovation **filter** having a transfer function of the form:F(z) = -az + 1 -az -10 where... ...
17 A method for enhancing periodicity as defined in claim 1 1, wherein said **filtering** comprises processing the innovation vector through an innovation **filter** having a transfer function of the form:F(Z) = 1 -CZ -1where a is...from said encoded wideband signal at least pitchcodebook parameters, innovative codebook parameters, and synthesis **filter**coefficients;b) an pitch codebook responsive to said pitch codebookparameters for producing a pitch... ...factor generator for calculating a periodicity factor related to thewideband signal, and said innovation **filter** for **filtering** the innovative codevector;e) a combiner circuit for combining said pitch codevector and saidinnovative codevector **filtered** by said innovation **filter** to thereby producesaid periodicity enhanced excitation signal; andf) a signal synthesis **filter** for **filtering** said periodicity enhancedexcitation signal in relation to said synthesis **filter** coefficients to thereby produce said synthesized wideband signal.
22 A decoder for producing a synthesized... ...decoder for producing a synthesized wideband signal as defined in claim 21, wherein said innovation **filter** has a transfer function of the form:F(z) = -az + 1 -azwhere a is... ...decoder for producing a synthesized wideband signal as defined inclaim 21, wherein said innovation **filter** has a transfer function of the form:F(z) = 1 -Ozwhere a... ...from said encoded wideband signal at least pitchcodebook parameters, innovative codebook parameters, and synthesis **filter** coefficients;b) an pitch codebook responsive to said pitch

codebookparameters for producing a pitch... ...andinnovative codevector to thereby produce an excitation signal; and0
e) a signal synthesis **filter** for **filtering** said excitation signal inrelation to said synthesis **filter** coefficients to thereby produce saidsynthesized wideband signal;the improvement comprising of a periodicity enhancing... ...generator for calculating a periodicity factor 5 related to the wideband signal, and said innovation **filter** for **filtering** the innovative codevector.
32 A decoder for producing a synthesized wideband signal as defined in... ...decoder for producing a synthesized wideband signal as defined in
claim 31, wherein said innovation **filter** has a transfer function of the form:F(z) = -az + 1 -azwhere a is...decoder for producing a synthesized wideband signal as defined in claim 31, wherein said innovation **filter** has a transfer function of the form:F(z) =1 -Cz -11 5 where... ...innovative codevector.
43 A cellular communication system as defined in claim 41, wherein said
innovation **filter** has a transfer function of the form:F(z)=-az+l -azwhere a is... ...innovative codevector.
47 A cellular communication system as defined in claim 41, wherein said
innovation **filter** has a transfer function of the form:F(z) = 1 -Czwhere a is a...53 A cellular mobile transmitter/receiver unit as defined in claim 51, whereinsaid innovation **filter** has a transfer function of the form:F(z) = -az + 1 -az -1where a... ...57 A cellular mobile transmitter/receiver unit as defined in claim 51, whereinsaid innovation **filter** has a transfer function of the form:F(Z) = i -CZ -11 5 where... ...codevector.0
63 A cellular network element as defined in claim 61, wherein said
innovation **filter** has a transfer function of the form:F(z) = -az + 1 -az5where a... ...innovative codevector.
67 A cellular network element as defined in claim 61, wherein said
innovation **filter** has a transfer function of the form:F(z) =1 -Cyz -1where c3 is...73 A bidirectional wireless communication sub-system as defined in claim71 , wherein said innovation **filter** has a transfer function of the form:F(z) = -az + 1 -az -1where a... ...77 A bidirectional wireless communication sub-system as defined in claim 71 wherein said innovation **filter** has a transfer function of the form:
F(Z) = 1 -CZ -15 where a...

20/3K/1 (Item 1 from file: 348) Links
EUROPEAN PATENTS

00387070
An optimized speech recognition method.
Optimal gestaltetes Verfahren zur Spracherkennung.
Procede optimise pour la reconnaissance de la parole.

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	Country	Number	Kind	Date	
Patent	EP	385124	A2	19900905	(Basic)
	EP	385124	A3	19920617	
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Application	EP	90102035		19900202	
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CLAIMS A	(English)		908
SPEC A	(English)		6233
CLAIMS B	(English)	EPAB95	407
CLAIMS B	(German)	EPAB95	391
CLAIMS B	(French)	EPAB95	465
SPEC B	(English)	EPAB95	6329
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Total Word Count (Document B) 7592			
Total Word Count (All Documents) 14733			

20/3K/2 (Item 2 from file: 348) Links
EUROPEAN PATENTS

00168596
Nonlinear signal processing in a speech recognition system.
Nichtlineare Signalverarbeitung in einem Spracherkennungssystem.
traitement non lineaire de signal dans un systeme de reconnaissance de la parole.

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	Country	Number	Kind	Date	
Patent	EP	179280	A2	19860430	(Basic)
	EP	179280	A3	19870715	
	EP	179280	B1	19900425	
Application	EP	85111905		19850920	
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Type	Pub. Date	Kind	Text

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